300-075 cisco

Number: 300-075
Passing Score: 800
Time Limit: 120 min

Sections
1. VCS Control
2. Collaboration Edge (VCS Expressway)
3. Configure CUCM Video Service Parameters
4. Describe and Implement Centralized Call Processing Redundancy
5. Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
6. Implement Call Control Discovery/ILS
7. Implement Video Mobility Features
8. Implement Bandwidth Management and Call Admission Control on CUCM
9. Mixed Questions

Exam A

QUESTION 1
How is a SIP trunk in Cisco Unified Communications Manager configured for SIP precondition?

A. The configuration is done by selecting a SIP precondition trunk for trunk type.
B. The configuration is automatically selected when RSVP is enabled for the location assigned to the trunk.
C. SIP precondition is configured by selecting E2E for RSVP over SIP on the default SIP profile assigned to the SIP trunk.
D. SIP precondition is configured by configuring a new SIP profile and selecting E2E for RSVP over SIP. The new SIP profile must then be assigned to the SIP trunk.

Correct Answer: D
Section: VCS Control
Explanation

Explanation/Reference:

QUESTION 2
Which device is needed to integrate H.320 into the Cisco video solution?

A. video gateway
B. MGCP gateway
C. H.323 gatekeeper
D. MCU

Correct Answer: C
Section: VCS Control
Explanation

Explanation/Reference:
Incorrect answer: A, B, D
Explanation: As with H.323 MCUs, H.320 gateways are provisioned in Cisco Unified CallManager as H.323 gateways, and then route patterns are configured to extend calls to these devices.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/4x/42video.html#wp1046523

QUESTION 3
Refer to the exhibit.

```
controller T1 1/0
 framing esf
 linecode b8zs
 pri-group timeslots 1-6,24

interface Serial1/0:23
 no ip address
 encapsulation hdlc
 isdn switch-type primary-ni
 isdn incoming-voice voice
 no cdp enable

interface Vlan120
 ip address 10.2.120.1 255.255.255.0
 h323-gateway voip bind srcaddr 10.2.120.1

voice-port 1/0:23

 dial-peer voice 123 pots
 incoming called-number .
 direct-inward-dial

 dial-peer voice 23000 voip
 destination-pattern 2...
 session target ipv4:10.1.5.11
 dtmf-relay h245-alphanumeric
 codec g711ulaw
 no vad
```

IT shows an H.323 gateway configuration in a Cisco Unified Communications Manager environment. An inbound PSTN call to this H.323 gateway fails to connect to IP phone extension 2001. The PSTN user hears a reorder tone. Debug isdn q931 on the H.323 gateway shows the correct called-party number as 5015552001. Which two configuration changes can correct this issue? (Choose two.)

A. Add port 1/0:23 to dial-peer voice 123 pots.

B. Ensure that the Significant Digits for inbound calls on the H.323 gateway configuration is 4.
C. Add a voice translation profile to truncate the number from 10 digits to 4 digits. Apply the voice translation profile to the Voice-port. The configuration field "Significant Digits for inbound calls" is left at default (All).
D. Add the command h323-gateway voip id on interface vlan120.
E. Change the destination-pattern on the dial-peer voice 23000 VoIP to 501501? and change the Significant Digits for inbound calls to 4.

Correct Answer: BE
Section: VCS Control
Explanation

Explanation/Reference:
Incorrect answer: A, C, D
Explanation: Choose the number of significant digits to collect, from 0 to 32. Cisco Unified Communications Manager counts significant digits from the right (last digit) of the number that is called.
Link: http://cisco.biz/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b06trunk.html

QUESTION 4
The following exhibit shows configs for H.323 gateway. You have been asked to implement TEHO from a remote branch office with area code 301 to the HQ office with area code 201 using Cisco Unified Communications Manager. The remote office has an MGCP gateway and the HQ office has an H.323 gateway. Once the call arrives at the HQ, it should break out to the PSTN as a seven-digit local call. Which statement about the route pattern is true?

```
dial-peer voice 7 pots
 destination-pattern 9[2-9]......
 port 1/0:23
!
dial-peer voice 11 pots
 destination-pattern 91[2-9]..[2-9]......
 prefix 1
 port 1/0:23
```

A. route pattern should be 91201.[2-9]XXXXXX with Discard Digit as Predot and Prefix 9
B. route pattern should be 91201.[2-9]XXXXXX with Discard Digit as Predot
C. route pattern should be 91201.[2-9]XXXXXX
D. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot
E. route pattern should be 9.1201[2-9]XXXXXX with Discard Digit as Predot and Prefix 9

Correct Answer: A
Section: VCS Control
Explanation

Explanation/Reference:
Incorrect answer: B, C, D, E
Explanation: Destination pattern is 91, HQ office area code is 201 .

QUESTION 5
How many active gatekeepers can you define in a local zone?

A. 1
B. 2
C. 5
D. 10
E. 15
F. unlimited

Correct Answer: A
Section: VCS Control
Explanation

Explanation/Reference:

QUESTION 6
Which two statements about the functionality of a gatekeeper are true? (Choose two.)

A. Cisco Unified Communications Manager has gatekeeper functionality built in.
B. Cisco Unified Communications Manager registers with a gatekeeper via SIP.

C. Cisco Unified Communications Manager registers with a gatekeeper via H.323.
D. A gatekeeper can enable CAC and AAR.
E. A gatekeeper can enable CAC, but not AAR.

Correct Answer: CE
Section: VCS Control
Explanation

Explanation/Reference:

QUESTION 7
Which option describes a function of SIP preconditions?

A. SIP preconditions enable end-to-end RSVP over an SIP trunk.
B. SIP preconditions enable RSVP between Cisco IP Phones.
C. SIP preconditions can be enabled in a gatekeeper.
D. SIP preconditions enable end-to-end RSVP for calls through the PSTN.

Correct Answer: A
Section: VCS Control
Explanation

Explanation/Reference:

QUESTION 8
Which statement about the function of a gatekeeper is true?

A. A gatekeeper improves call routing between servers within a single Cisco Unified Communications Manager cluster.
B. A gatekeeper can replace the dial plan of a Cisco Unified Communications Manager cluster.
C. A gatekeeper can simplify the dial plan between many different Cisco Unified Communications Manager clusters.
D. Gatekeepers can be implemented to deploy RSVP-based CAC.

Correct Answer: C
Section: VCS Control
Explanation

Explanation/Reference:

QUESTION 9
For which VoIP protocol does a gatekeeper provide address translation and control access?

A. H.323
B. SIP
C. Skinny
D. H.248

Correct Answer: A
Section: VCS Control
Explanation

Explanation/Reference:

QUESTION 10
Which CAC configuration on a gatekeeper restricts to 10 G.711 audio calls?

A. Use the command bandwidth 10.
B. Use the command bandwidth 1280.
C. Use the command bandwidth 160.
D. Use the command bandwidth session 10.

Correct Answer: B
Section: VCS Control
Explanation

Explanation/Reference:

QUESTION 11
If the device pool in the phone record does not match the device pools in the matching subnet, what will the system consider the phone to be?

A. roaming
B. unregistered
C. unknown
D. new device

Correct Answer: A
Section: VCS Control
Explanation

Explanation/Reference:

QUESTION 12
What user profile is used to define the settings for a user on login?

A. Device Profile
B. Group Profile
C. Pool Profile
D. Specific Profile

Correct Answer: A
Section: VCS Control
Explanation

Explanation/Reference:

QUESTION 13
Which statement about technology implementation strategy is true?

A. Cisco Unified Communications Manager Express can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.
B. Cisco Unified Communications Manager Express in SRST mode can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.
C. SRST can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.
D. SRST and MGCP fallback can be configured to function with no Cisco Unified Communications Manager cluster in the enterprise.

Correct Answer: A
Section: Collaboration Edge (VCS Expressway)
Explanation

Explanation/Reference:

QUESTION 14
Which commands are needed to configure Cisco Unified Communications Manager Express in SRST mode?

A. telephony-service and srst mode
B. telephony-service and moh
C. call-manager-fallback and srst mode
D. call-manager-fallback and voice-translation

Correct Answer: A
Section: Collaboration Edge (VCS Expressway)
Explanation

Explanation/Reference:

QUESTION 15
What command is used to map internal extensions to the corresponding E.164 PSTN number when using Cisco Unified Communications Manager Express in SRST mode?

A. ephone-dn
B. dialplan-pattern
C. number
D. number-e.164
E. ephone-transnumber

Correct Answer: B
Section: Collaboration Edge (VCS Expressway)
Explanation

Explanation/Reference:

QUESTION 16
Which statement about enrollment in the IP telephony PKI is true? (Source. Understanding Cisco IP Telephony Authentication and Encryption Fundamentals)

A. CAPF enrollment supports the use of authentication strings.
B. The CAPF itself has to enroll with the Cisco CTL client.
C. LSCs are issued by the Cisco CTL client or by the CAPF.
D. MICs are issued by the CAPF itself or by an external CA.

Correct Answer: A
Section: Collaboration Edge (VCS Expressway)
Explanation

Explanation/Reference:
Incorrect answer: B, C, D
Explanation: The CAPF enrollment process is as follows:
1. The IP phone generates its public and private key pairs.
2. The IP phone downloads the certificate of the CAPF and uses it to establish a TLS session with the CAPF.
3. The IP phone enrolls with the CAPF, sending its identity, its public key, and an optional authentication string.
4. The CAPF issues a certificate for the IP phone signed with its private key.
5. The CAPF sends the signed certificate to the IP phone.

Link: http://my.safaribooksonline.com/book/certification/cipt/9781587052613/understanding-cisco-ip-telephony-authentication-and-encryption-fundamentals/584.

QUESTION 17
In what Cisco solution is Simple Network-Enabled Auto Provision technology used?

A. Cisco Unified Gateway Duplication
B. Cisco Unified CallManager Redundancy
C. Cisco Unified SRST
D. Cisco Unified Call Survivability

Correct Answer: C
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:
Incorrect answer: A, B, D
Explanation: When the system automatically detects a failure, Cisco Unified SRST uses Simple Network Auto Provisioning (SNAP) technology to auto-configure a branch office router to provide call processing for the Cisco Unified IP phones that are registered with the router
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmesrst.html

QUESTION 18
While configuring Call Survivability in Cisco Unified Communications Manager, what step is mandatory to reach remote sites while in SRST mode?

A. Enable Cisco Remote Site Reachability.
B. Configure CFUR.
C. Enable the SRST checkbox in the MGCP gateway.
D. Configure the H.323 gateway for SRST in Cisco Unified Communications Manager.
E. Enable the Failover Service parameter.

Correct Answer: B
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:
Incorrect answer: A, C, D, E
Explanation: Call Forward Unregistered (CFUR) functionality provides the automated rerouting of calls through the PSTN when an endpoint is considered unregistered due to a remote WAN link failure
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/8x/models.html

QUESTION 19
While operating in SRST, what is needed to route calls outside of the remote site location to the PSTN?

A. SIP trunk
B. CallManager route patterns
C. translation patterns
D. POTS dial peers
E. VOIP dial peers

Correct Answer: D

Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:
Incorrect answer: A, B, C, E
Explanation: in time of srst configuration on router, please configure a dial-peer so that call flow in SRST mode.
Link:

QUESTION 20
When using Cisco Unified Communications Manager Express in SRST mode, how many multicast music on hold streams can be utilized by the system at any given time?

A. 3
B. 6
C. 2
D. 4
E. 1
F. 5

Correct Answer: B
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 21
To preserve analog calls in an MGCP switchback event, which three commands must be configured in the MGCP fallback router? (Choose three.)

A. h323
B. mgcp-switchback-graceful
C. voice service voip
D. mgcp-graceful
E. preserve-h323
F. no h225 timeout keepalive

Correct Answer: ACF
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:
Incorrect answer: B, D, E
Explanation: these additional command for call preservation when using MGCP fallback:
voice service voip
h323
no h225 timeout keepalive
Reference. CCNP Voice CIPT2 642-457 Quick Reference, 2nd Edition

QUESTION 22
Which two locations are the best locations that an end user can use to determine if an IP phone is working in SRST mode? (Choose two.)

A. Cisco Unified Communications Manager Administration
B. IP phone display
C. Cisco Unified SRST Router
D. Cisco Unified MGCP Fallback Router
E. physical IP phone settings

Correct Answer: BE
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:
Incorrect answer: A, C, D
Explanation: IP Phone display and Physical phone IP settings are two locations were an end user can determine if an IP phone is working in SRST mode.
Link: http://my.safaribooksonline.com/book/telephony/1587050757/survivable-remote-site-telephony-srst/529

QUESTION 23
What is the fastest way for an engineer to test the implementation of SRST in a production environment?

A. Shut down the Cisco Unified Communications Manager Servers.
B. Shut down the switch ports connected to the Cisco Unified Communications Manager Servers.
C. Add a null route to the publisher Cisco Unified Communications Manager at the remote router. Remove the null route when the operation is verified.
D. Unplug the IP phones from their switch ports.
E. Verification is not needed.

Correct Answer: C
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 24
DRAG DROP

Select and Place:

Click and drag the minimum Cisco Unified SRST configuration steps on the left to the spaces on the right. Not all spaces on the right are used.

- ccm-manager fallback-mgcp
- call-manager-fallback
- application
- ip source-address 10.5.1.1 port 2000
- service alternate default
- max-ephones 3
- max-dn 6
- limit-dn 7965 2
- keepalive 20

Correct Answer:

Click and drag the minimum Cisco Unified SRST configuration steps on the left to the spaces on the right. Not all spaces on the right are used.

- ccm-manager fallback-mgcp
- application
- service alternate default
- limit-dn 7965 2
- keepalive 20

- call-manager-fallback
- ip source-address 10.5.1.1 port 2000
- max-ephones 3
- max-dn 6

Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 25
Which solution is needed to enable presence and extension mobility to branch office phones during a WAN failure?

A. SRST with MGCP fallback
B. SRST without MGCP fallback
C. Cisco Unified Communications Manager Express in SRST mode
D. SRST with VoIP dial peers to Cisco Unified Communications Manager Express

Correct Answer: C
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 26
Which two options enable routers to provide basic call handling support for Cisco Unified IP Phones if they lose connection to all Cisco Unified Communications Manager systems? (Choose two.)

A. SCCP fallback
B. Cisco Unified Survivable Remote Site Telephony
C. Cisco Unified Communications Manager Express
D. MGCP fallback
E. Cisco Unified Communications Manager Express in SRST mode

Correct Answer: BE
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 27
Which two statements about remote survivability are true? (Choose two.)

A. SRST supports more Cisco IP Phones than Cisco Unified Communications Manager Express in SRST mode.
B. Cisco Unified Communications Manager Express in SRST mode supports more Cisco IP Phones than SRST.
C. MGCP fallback is required for ISDN call preservation.
D. MGCP fallback functions with SRST.

Correct Answer: AD
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 28
You are the Cisco Unified Communications Manager in Certpaper.com. You use a remote site MGCP gateway to provide redundancy when connectivity to the central Cisco Unified Communications Manager cluster is lost. How to enable IP phones to establish calls to the PSTN when they have registered with the gateway? (Choose three.)

A. POTS dial peers must be added to the gateway to route calls from the IP phones to the PSTN.
B. The default service must be enabled globally.
C. The command ccm-manager mgcp-fallback must be configured.
D. COR needs to be configured to disallow outbound calls.

Correct Answer: ABC
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:
Incorrect answer: D
Explanation: Class of restriction: Cisco Unified Communications Manager Business Edition 3000 supports class of service (CoS) with respect to geographic reach as follows:
– Campus
– Local
– National
– International
– Emergency services

- Call waiting
- Default ringtones
- MoH
- Speed dials: Single-button, not BLF

Link: http://www.cisco.com/en/US/prod/collateral/voicesw/ps6788/vcallcon/ps11370/data_sheet_c78-651909.html

QUESTION 29
Which technologies provide remote-site redundancy for Cisco IP Phones during a WAN failure?

A. SRST and MGCP fallback
B. SRST and TEHO
C. TEHO and MGCP fallback
D. SRST and AAR

Correct Answer: A
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 30
Which remote-site redundancy technology fails over to POTS dial peers from the Cisco Unified Communications Manager dial plan during a WAN failure?

A. MGCP fallback
B. H.323 fallback
C. SCCP fallback
D. SIP fallback

Correct Answer: A
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 31
How does the system intelligently shift call processing upon restoration of WAN connectivity?

A. automatically back to the primary Cisco Unified Communications Manager cluster
B. manually back to the primary Cisco Unified Communications Manager cluster
C. automatically back to the secondary Cisco Unified Communications Manager cluster
D. manually back to the secondary Cisco Unified Communications Manager cluster

Correct Answer: A
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 32
Which option configures call preservation for H.323-based SRST mode?

A. voice service voip h323 call preserve
B. call preservation not possible with H.323
C. call-manager-fallback preserve-call
D. dial-peer voice 1 voip call preserve

Correct Answer: A
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 33
Which configuration command disables the secondary dial tone on the branch office ISR for users calling from the PSTN into the branch office during a WAN failure?

A. direct-inward-dial
B. voice translation-rule
C. incoming called-number
D. application

Correct Answer: A
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 34
Refer to the exhibit.

```
dial-peer voice 901 pots
 destination-pattern 9011T
 port 1/0:23
```

Which configuration change is needed to enable NANP international dialing during MGCP fallback?

A. Change the dial peer to dial-peer voice 901 voip.
B. Change the dial peer to dial-peer voice 9011 pots.
C. Add the command prefix 011 to the dial peer.
D. Add the command prefix 9011 to the dial peer.

Correct Answer: C
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 35
Which three commands are mandatory to implement SRST for five Cisco IP Phones? (Choose three.)

A. call-manager-fallback
B. max-ephones
C. keepalive
D. limit-dn
E. ip source-address

Correct Answer: ABE
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 36
A Cisco Unified Communications Manager cluster is installed in headquarters only.

How can international calls be blocked while national calls are allowed for branch office Cisco IP Phones during a WAN failure?

A. Configure CSS and partitions in Cisco Unified Communications Manager and apply the CSS and partitions to the SRST ISR.
B. Configure CSS and partitions in the SRST ISR.
C. Configure COR in the SRST ISR.
D. Configure voice translations in the SRST ISR.

Correct Answer: C
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 37
Which command can be used to manually send the MGCP gateway to register with the secondary Cisco Unified Communications Manager server?

A. ccm-manager switchover-to-backup
B. mgcp use backup
C. ccm-manager register backup
D. not supported

Correct Answer: A
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 38
Which command is needed to utilize local dial peers on an MGCP-controlled ISR during an SRST failover?

A. ccm-manager fallback-mgcp
B. telephony-service
C. dialplan-pattern
D. isdn overlap-receiving
E. voice-translation-rule

Correct Answer: A
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 39
Which option is a benefit of implementing CFUR?

A. CFUR is designed to initiate TEHO to reduce toll charges.
B. CFUR can prevent phones from unregistering.
C. CFUR can reroute calls placed to a temporarily unregistered destination phone.
D. CFUR eliminates the need for COR on an ISR.

Correct Answer: C
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 40
This is the configuration on the voice gateway:

telephony-service
max-ephones 30
max-dn 60 preference 0
srst mode auto-provision all

srst dn line-mode dual
srst dn template 3
srst ephone description srst fallback auto-provision phone
srst ephone template 5

Which ephone-dn would be expected upon activation of SRST?

A. ephone-dn 1 dual-line
number 7001
description 7001
name 7001
ephone-dn-template 5
This DN is learned from srst fallback ephones
B. ephone-dn 1 dual-line
number 7001
description 7001
name 7001
ephone-dn-template 3
This DN is learned from srst fallback ephones
C. ephone-dn 1
number 7001
description 7001
name 7001
ephone-dn-template 5
This DN is learned from srst fallback ephones
D. ephone-dn 1
number 7001
description 7001
name 7001
ephone-dn-template 3
This DN is learned from srst fallback ephones

Correct Answer: A
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 41
Which ability does the Survivable Remote Site Telephony feature provide?

A. a means to allow the local site to continue to send and receive calls in the event of a WAN failure
B. a means to route calls on-net through other sites during high utilization periods
C. a method that allows for backup calls in the event that your gateway fails
D. the ability to force a call out of a certain trunk when the Cisco Unified Communications Manager is being upgraded

Correct Answer: A
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 42
When implementing a dial plan for multisite deployments, what must be present for SRST to work successfully?

A. dial peers that address all sites in the multisite cluster
B. translation patterns that apply to the local PSTN for each gateway
C. incoming and outgoing COR lists
D. configuration of the gateway as an MGCP gateway

Correct Answer: B
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 43
What component acts as a user agent for both ends of a SIP call while Cisco Unified SIP SRST is providing failover during a WAN outage?

A. B2BUA
B. SIP server
C. SIP proxy
D. SIP SRST router
E. SIP registrar

Correct Answer: A

Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 44
Which two configurations are needed to implement SRST in Cisco Unified Communications Manager? (Choose two.)

A. SRST Gateway setting in Cisco Unified Communications Manager
B. SRST Reference configured in Cisco Unified Communications Manager
C. Device Pool SRST Reference setting
D. Call Manager Group setting
E. Cisco Unified Communications Locations setting

Correct Answer: BC
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 45
Which of the following are two functions that ensure that the telephony capabilities stay operational in the remote location Cisco Unified SRST router? (Choose two)

A. Automatically detecting a failure in the network.
B. Initiating a process to provide call-processing backup redundancy.
C. Notifying the administrator of an issue for manual intervention.
D. Proactively repairing issues in the voice network.

Correct Answer: AB
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

QUESTION 46
Which three of the following are steps in configuring MGCP Fallback and Cisco Unified SRST? (Choose three)

A. Define the SRST reference for phones in the Device Pool configuration
B. Enable and configure the MGCP fallback and Cisco Unified SRST features on the IOS gateways.
C. Implement a simplified SRST dial plan on the remote-site-gateways to ensure connectivity for remote-site phones in SRST mode.
D. Enable SIP trunking between both remote and hub sites to provide mesh coverage.
E. Define the SRST reference in the configuration on the IP Phones.
F. Enable and configure the MGCP fallback on the IOS gateway but not Cisco Unified SRST since it is enabled automatically.

Correct Answer: ABC
Section: Describe and Implement Centralized Call Processing Redundancy
Explanation

Explanation/Reference:

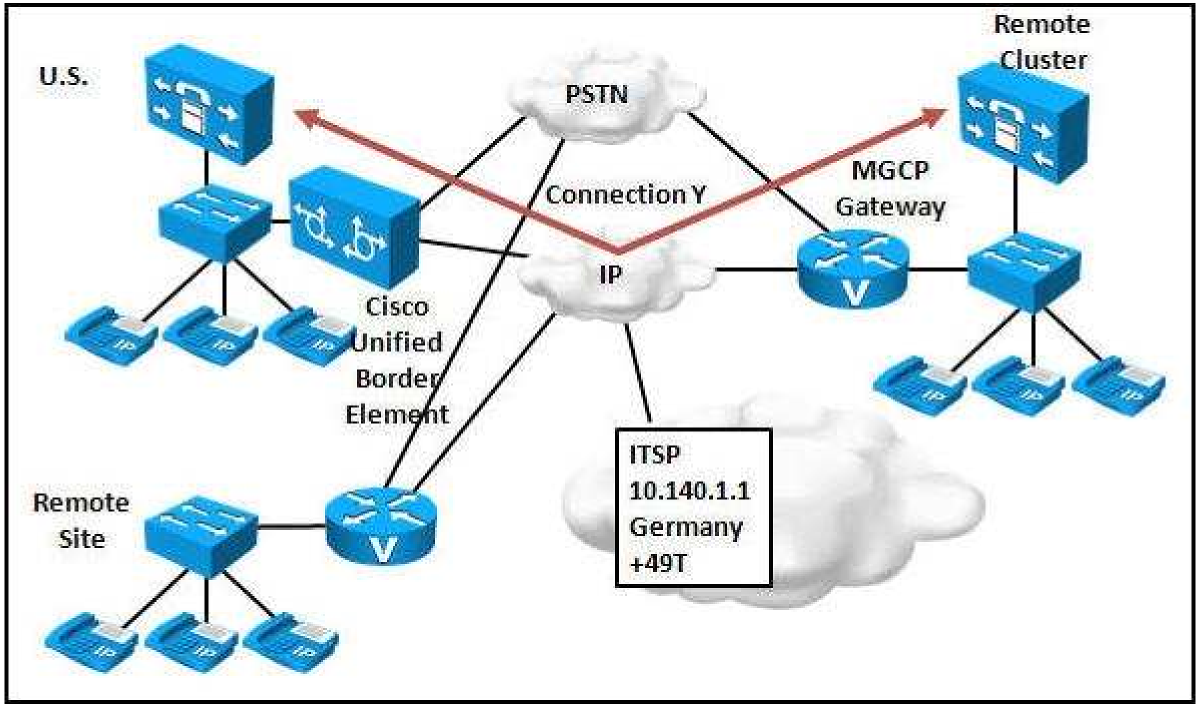
QUESTION 47
Which method can be used to address variable-length dial plans?

A. Overlap sending and receiving.
B. Add a prefix for all calls that are longer than 10-digits long
C. Use nested translation patterns to eliminate inter-digit timeout
D. Use the @macro on the route pattern
E. Use MGCP gateways, which support variable-length dial plans

Correct Answer: A
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:
Incorrect answer: B, C, D, E
Explanation: If the dial plan contains overlapping patterns, Cisco Unified Communications Manager does not route the call until the interdigit timer expires (even if it is possible to dial a sequence of digits to choose a current match). Check this check box to interrupt interdigit timing when Cisco Unified Communications Manager must route a call immediately.
By default, the Urgent Priority check box displays as checked. Unless your dial plan contains overlapping patterns or variable length patterns that contain!, Cisco recommends that you do not uncheck the check box.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsintrcm.html

QUESTION 48
Refer to the exhibit.



Which trunks would be most suitable for Connection Y?

A. an H.225 trunk (gatekeeper-controlled)
B. intercluster trunk (gatekeeper-controlled)
C. a SIP trunk on the U.S. cluster and an intercluster trunk on the remote cluster
D. intercluster trunk (nongatekeeper-controlled)
E. No extra connections are required. As long as IP connectivity exists, you need only configure a route pattern for each site. The Cisco Unified Communications Manager will automatically forward the calls over the WAN if the destination directory number is not registered locally.

Correct Answer: D
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 49
Which two features require or may require configuring a SIP trunk? (Choose two.)

A. SIP gateway
B. Call Control Discovery between a Cisco Unified Communications Manager and Cisco Unified Communications Manager Express
C. Cisco Device Mobility
D. Cisco Unified Mobility
E. registering a SIP phone

Correct Answer: AB
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:
Incorrect answer: C, D, E
Explanation: All protocols require that either a signaling interface (trunk) or a gateway be created to accept and originate calls. Device mobility allows Cisco Unified Communications Manager to determine whether the phone is at its home location or at a roaming location. Cisco Unified Mobility gives users the ability to redirect incoming IP calls from Cisco Unified Communications Manager to different designated phones, such as cellular phones.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08sip.html#wpxref77849

QUESTION 50
A Cisco 3825 needs to be added in Cisco Unified Communications Manager as an H.323 gateway. What should the gateway type be?

A. H.323 gateway
B. Cisco 3825
C. Cisco 3800 series router. The specific model will be selected after the Cisco 3800 is selected.
D. The gateway can be added either as an H.323 gateway or Cisco 3800 series router.
E. The gateway can be added either as an H.323 gateway or Cisco 3825 series router.

Correct Answer: A
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 51

Which statement best describes globalized call routing in Cisco Unified Communications Manager?

A. All incoming calling numbers on the phones are displayed as an E.164 with the + prefix.
B. Call routing is based on numbers represented as an E.164 with the + prefix format.
C. All called numbers sent out to the PSTN are in E.164 with the + prefix format.
D. The CSS of all phones contain partitions assigned to route patterns that are in global format.
E. All phone directory numbers are configured as an E.164 with the + prefix.

Correct Answer: B
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:
Incorrect answer: A, C, D, E
Explanation: For the destination to be represented in a global form common to all cases, we must adopt a global form of the destination number from which all local forms can be derived. The + sign is the mechanism used by the ITU's E.164 recommendation to represent any PSTN number in a global, unique way. This form is sometimes referred to as a fully qualified PSTN number.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/7x/dialplan.html#wp1153205

QUESTION 52
When an incoming PSTN call arrives at an MGCP gateway, how does the calling number get normalized to a global E.164 number with the + prefix in Cisco Unified Communications Manager?

A. Normalization is done using translation patterns.
B. Normalization is done using route patterns.
C. Normalization is done using the gateway incoming called party prefixes based on number type.
D. Normalization is done using the gateway incoming calling party prefixes based on number type.
E. Normalization is achieved by local route group that is assigned to the MGCP gateway.

Correct Answer: D
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:
Incorrect answer: A, B, C, E
Explanation: Configuring calling party normalization alleviates issues with toll bypass where the call is routed to multiple locations over the IP WAN. In addition, it allows Cisco Unified Communications Manager to distinguish the origin of the call to globalize or localize the calling party number for the phone user.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fscallpn.html

QUESTION 53
When an incoming PSTN call arrives at an MGCP gateway, how does the called number get normalized to an internal directory number in Cisco Unified Communications Manager?

A. Normalization is done by configuring the significant digits for inbound calls on the MGCP gateway.
B. Normalization is done using route patterns.
C. Normalization is done using the gateway incoming called party prefixes based on number type.
D. Normalization is done using the gateway incoming calling party prefixes based on number type.
E. Normalization is achieved by local route group that is assigned to the MGCP gateway.

Correct Answer: A
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

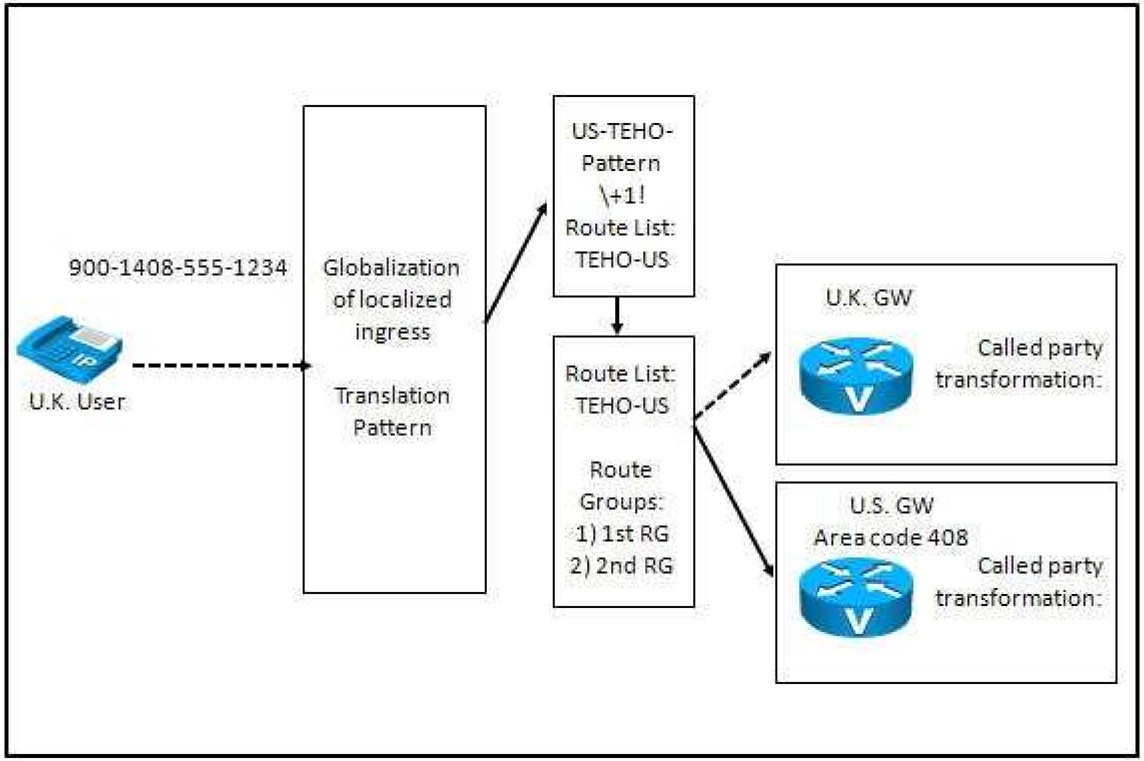
QUESTION 54
Which process can localize a global E.164 with + prefix calling numbers for inbound calls to an IP phone so that users see the calling number in a local format?

A. Calling number localization is done using translation patterns.
B. Calling number localization is done using route patterns.
C. Calling number localization is done by configuring a calling party transformation CSS at the phone.
D. Calling number localization is done by configuring a calling party transformation CSS at the gateway.
E. Calling number localization is done by configuring the phone directory number in a localized format.

Correct Answer: C
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 55
Refer to the exhibit.



The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. To match the US-TEHO pattern \+!, how should the translation pattern be configured?

A. 9001.4085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +

B. 9.0014085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +1

C. 900.14085551234 with the Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +1

D. 900.14085551234 with the Called Party Transformation:

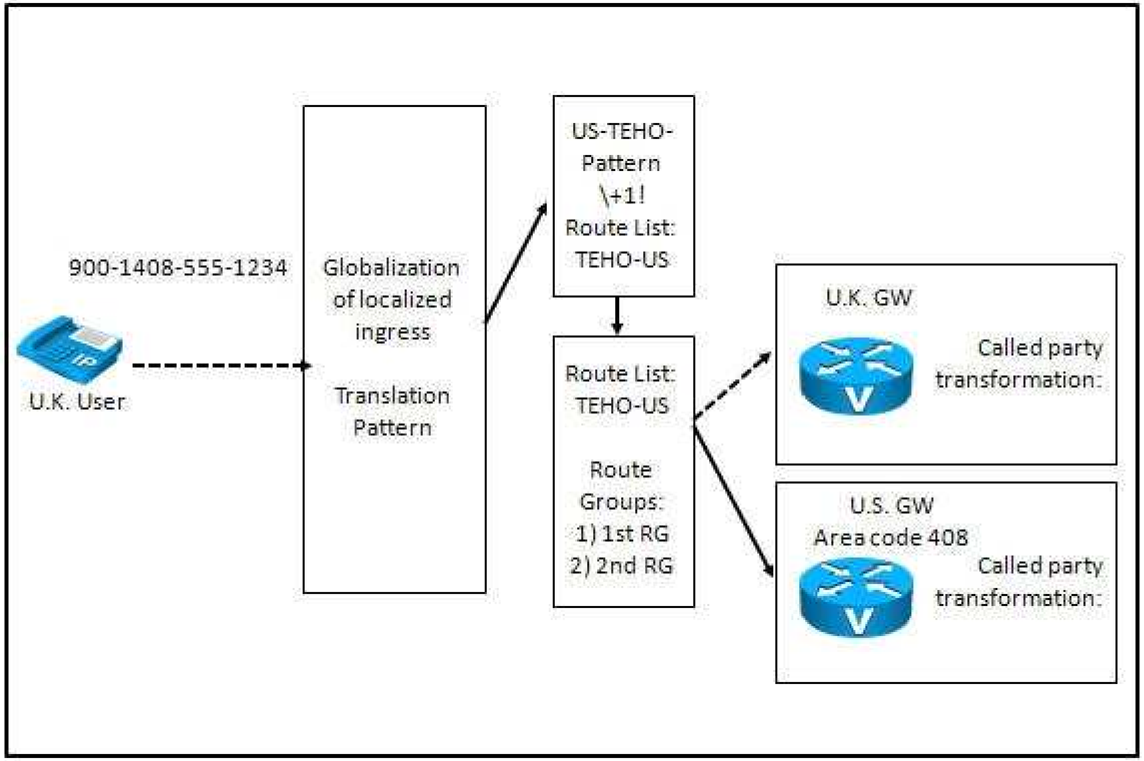
Discard Digits PreDot
Prefix Digits Outgoing Calls: +

E. 001.4085551234 with the Called Party Transformation:
Prefix Digits Outgoing Calls: +

Correct Answer: D
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:
Incorrect answer: A, B, C
Explanation: The PSTN access code for the UK is 9, International call code is 001, The international escape character, +, signifies the international access code in a complete E.164 number format
Link: http://www.ciscosystems.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a03rp.html

QUESTION 56
Refer to the exhibit.



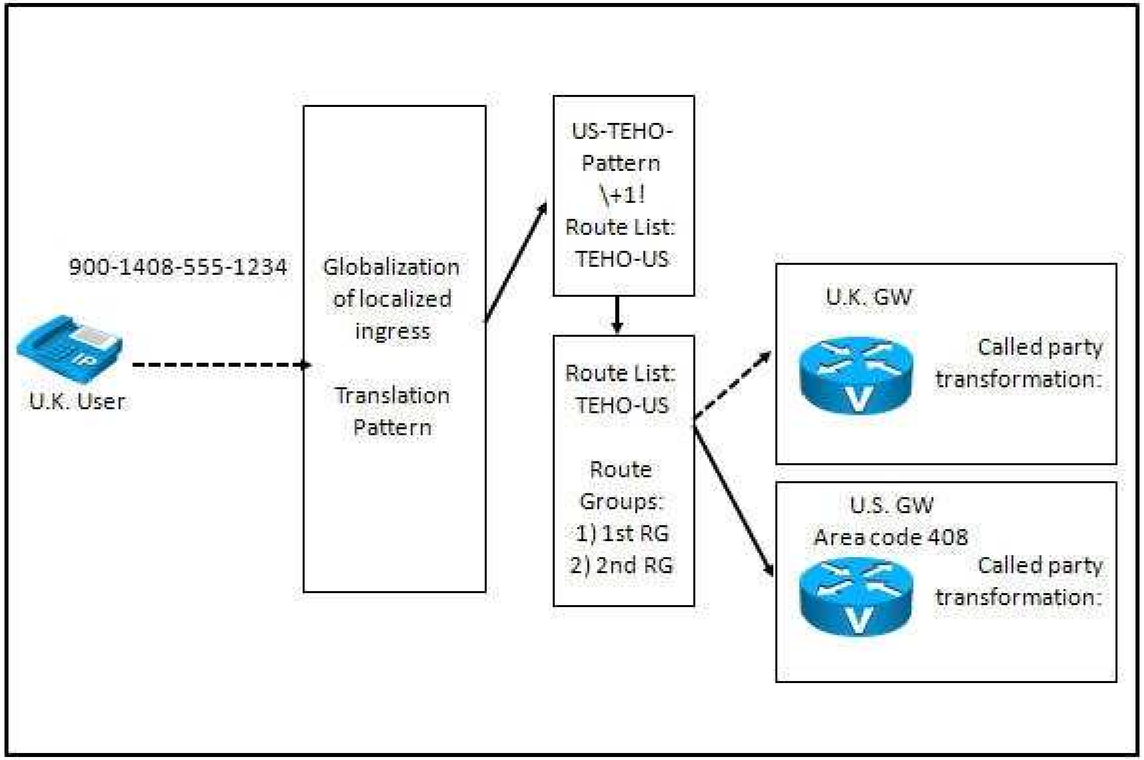
The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. What should the TEHO-US route list configuration consist of?

A. First route group should point only to the U.K. gateway. The second route group should point to the U.S. gateway.
B. First route group should be only the local route group. The second route group should point to the U.S. gateway.
C. First route group should point only to the U.S. gateway. The second route group should be the local route group.
D. The TEHO-US route list should contain only the local route group. The globalized configuration means that the appropriate gateway will be selected automatically.
E. The \+! route pattern should point directly to the U.S. gateway.

Correct Answer: C
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:
Incorrect answer: A, B, D
Explanation: The route group points to one or more gateways and can choose the gateways for call routing based on preference. The route group can serve as a trunk group by directing all calls to the primary device and then using the secondary devices when the primary is unavailable. One or more route lists can point to the same route group.
Link: http://www.ciscosystems.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08gw.html#wp1167274

QUESTION 57
Refer to the exhibit.



The exhibit shows centralized Cisco Unified Communications Manager configuration components for TEHO calls to U.S. area code 408 from the U.K. The PSTN access code for the U.K. is 9 and 001 for international calls to the U.S. Assuming the PSTN does not accept globalized numbers with + prefix. What should the Called Party Transformation Pattern at the U.S. gateway be configured as?

A. \+.! with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: +

B. \+1.! with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: None

C. \+408.! with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: 1

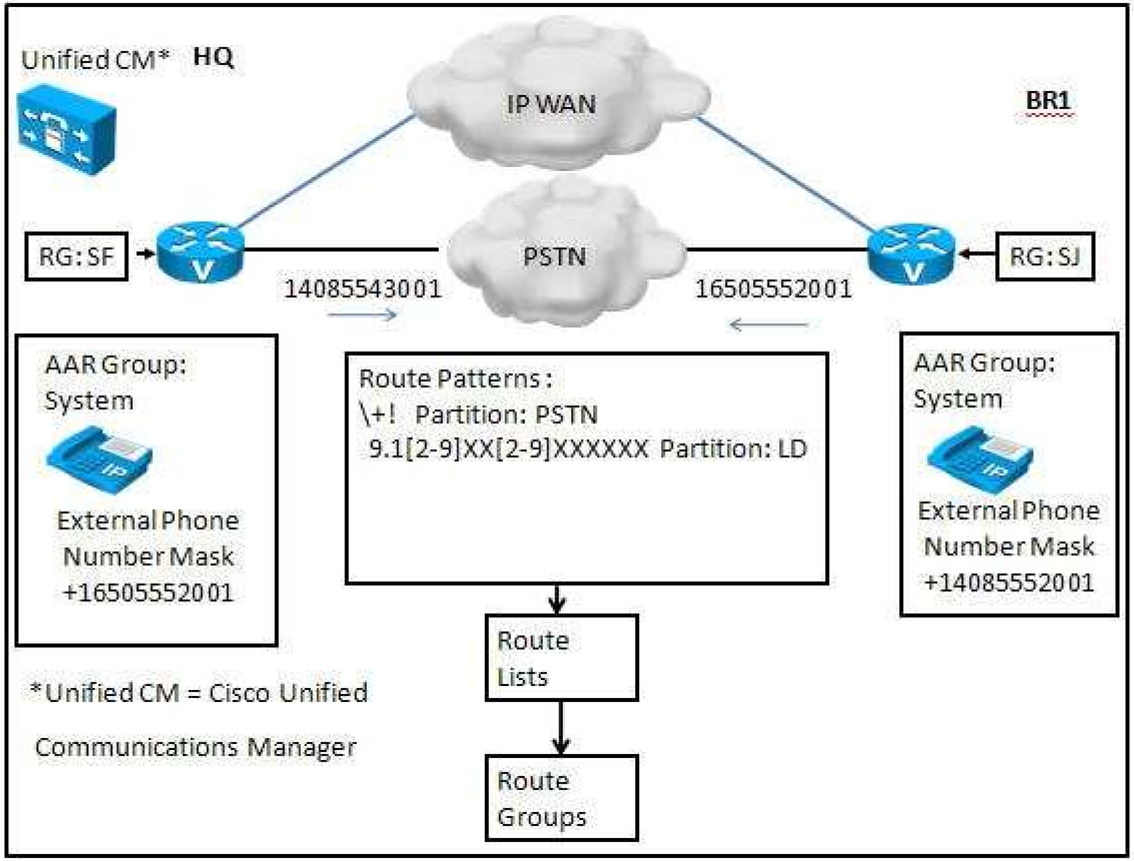
D. \+1408.! with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: None

E. \+1.408! with the following Called Party Transformation:
Discard Digits PreDot
Prefix Digits Outgoing Calls: None

Correct Answer: D
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 58
Refer to the exhibit.



The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number. Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit.

What should the AAR group prefix be?

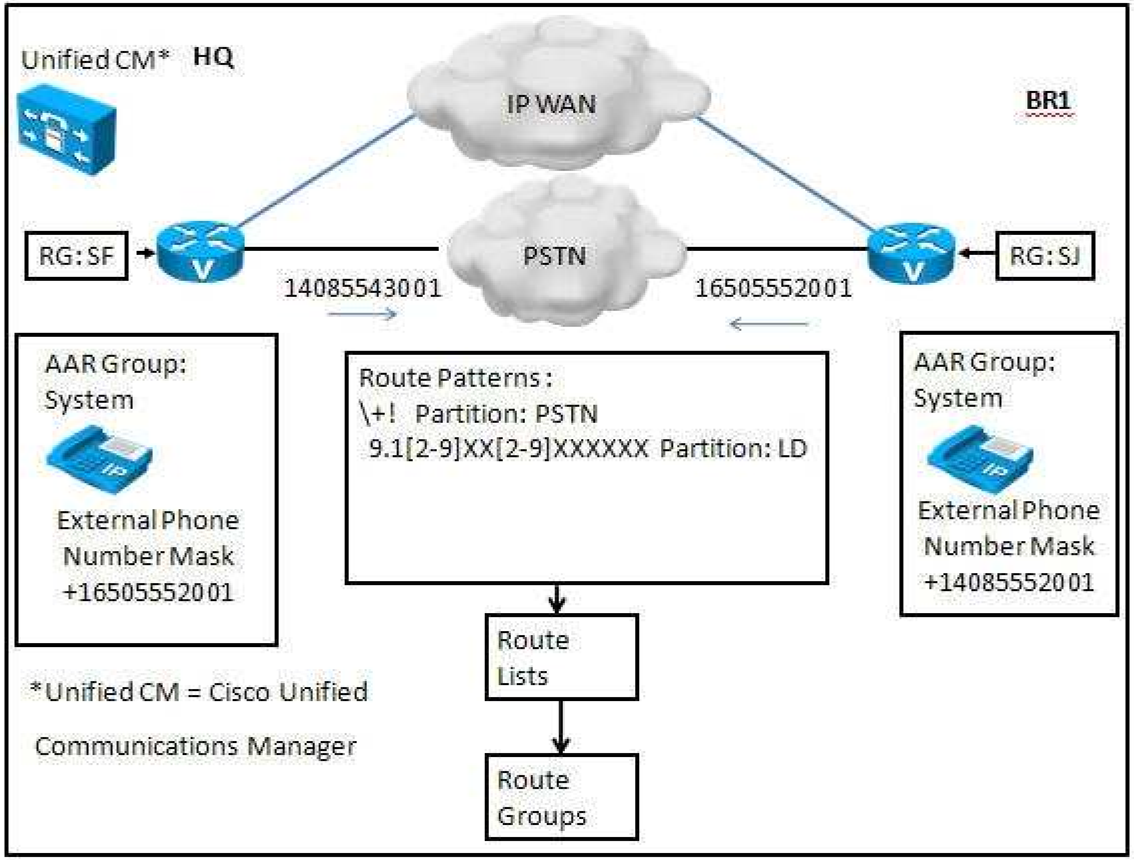
A. 9
B. 91
C. none
D. +

E. +1

Correct Answer: C
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 59
Refer to the exhibit.



The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number.

Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit.

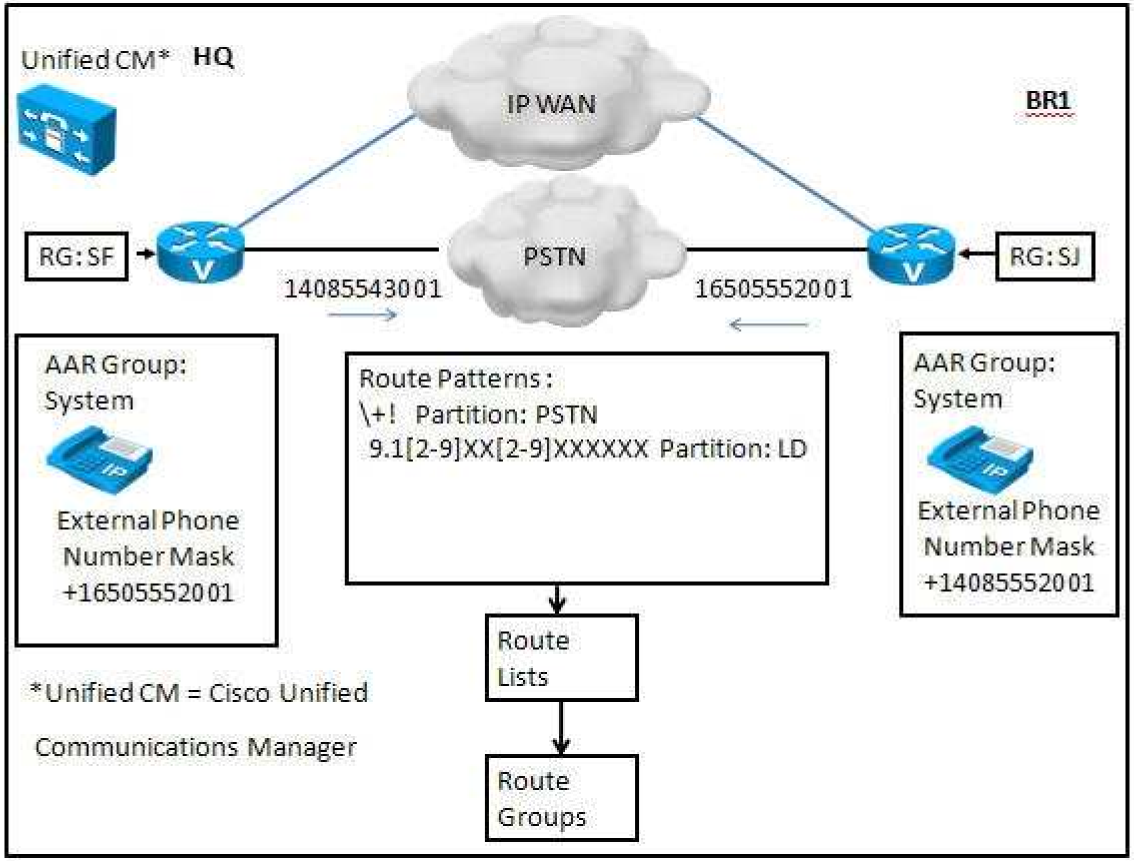
Which partition should be configured in the AAR CSS applied at the phones?

A. PSTN partition
B. LD partition
C. The HQ AAR CSS must include a partition assigned to route pattern 91408XXXXXXX. The BR1 AAR CSS must include a partition assigned to route pattern 91650XXXXXXX.
D. AAR CSS must contain translation pattern 9.1[2-9]XX[2-9]XXXXXX for each site that must be globalized. Otherwise the called numbers will not be localized at the egress gateway.

Correct Answer: A
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 60
Refer to the exhibit.



The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number.

Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit.

How many route lists and route groups should be configured for AAR at a minimum?

A. a single route list with a local route group for each site
B. two route lists and two route groups for each site
C. a single route list and four route groups for each site

D. None. The AAR CSS can point directly to the route pattern.

Correct Answer: A
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 61
Refer to the exhibit.

RTP Phone Device Configuration	Partitions
Device CSS	RTP_Emergency ALL_Phones
AAR CSS	RTP_LongDistance

RTP Phone DN Configuration	Partitions
Line CSS	RTP_Local RTP_LongDistance RTP_International
AAR Group	AAR

U.K. User Device Profile	Partitions
Line CSS	U.K_Emergency ALL_Phones
AAR Group	AAR

Partition	Route Pattern
RTP_Emergency	9.911
RTP_Local	9.[2-9]XXXXXX
RTP_LongDistance	9.1[2-9]XX[2-9]XXXXXX
RTP_International	9.011!#
U.K_Emergency	0.000
U.K_PSTN	9.!

Assume a centralized Cisco Unified Communications deployment with the headquarters in the U.K, and remote site in RTP. All route patterns are assigned a route list that points to the local route group. Local route groups have been configured on the U.K and RTP device pools. A U.K. user logs onto an RTP phone using the

Cisco Extension Mobility feature and places an emergency call to 0000. Which statement about the emergency call is true?

A. The call will match the U.K_Emergency route pattern partition and will egress at the RTP gateway.
B. The call will match the U.K_Emergency route pattern partition and will egress at the U.K. gateway.
C. The call will match the RTP_Emergency route pattern partition and will egress at the U.K. gateway.
D. The call will match the RTP_Emergency route pattern partition and will egress at the RTP gateway.
E. The call will fail.

Correct Answer: A
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 62
Refer to the following exhibit.

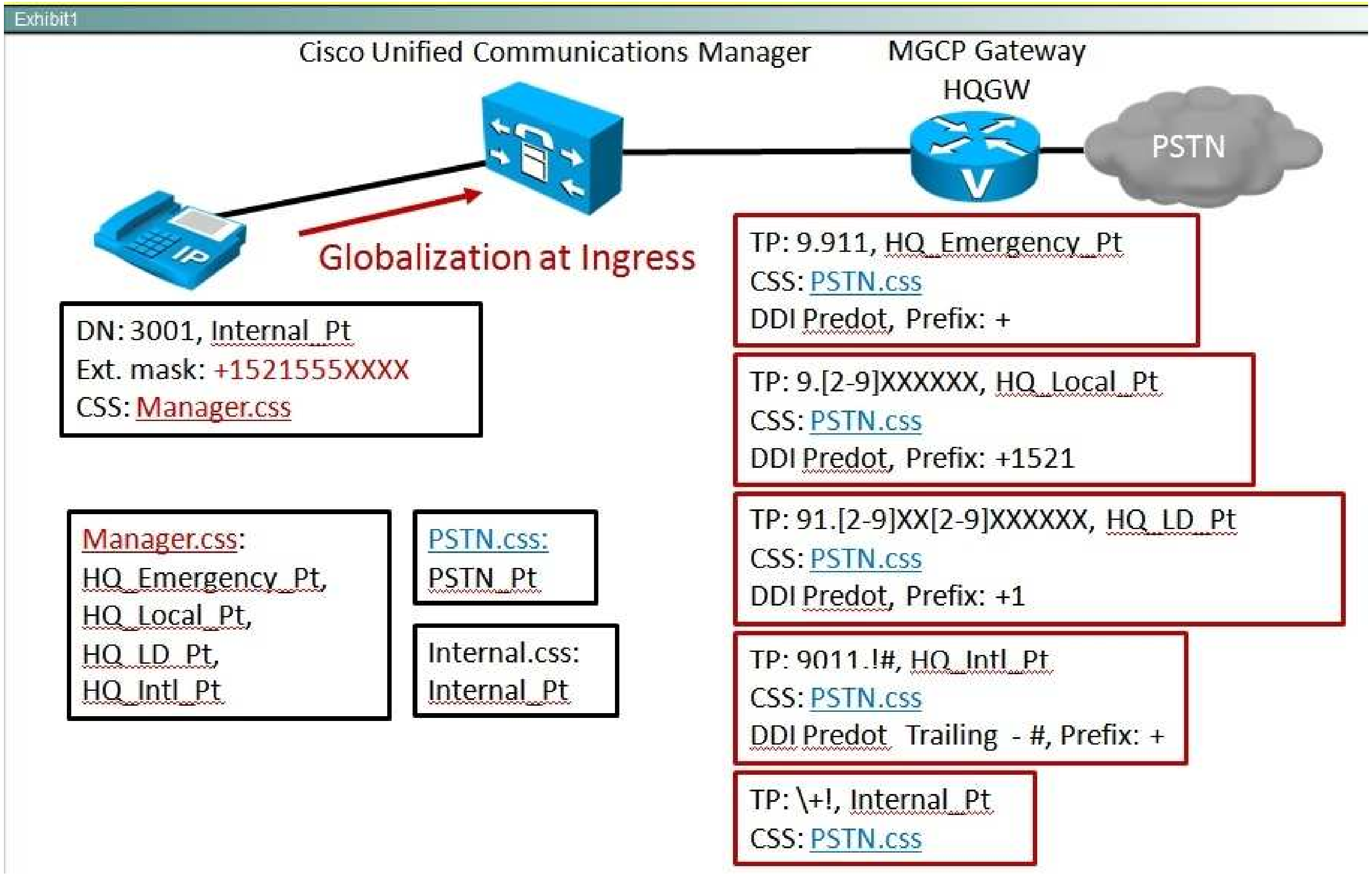
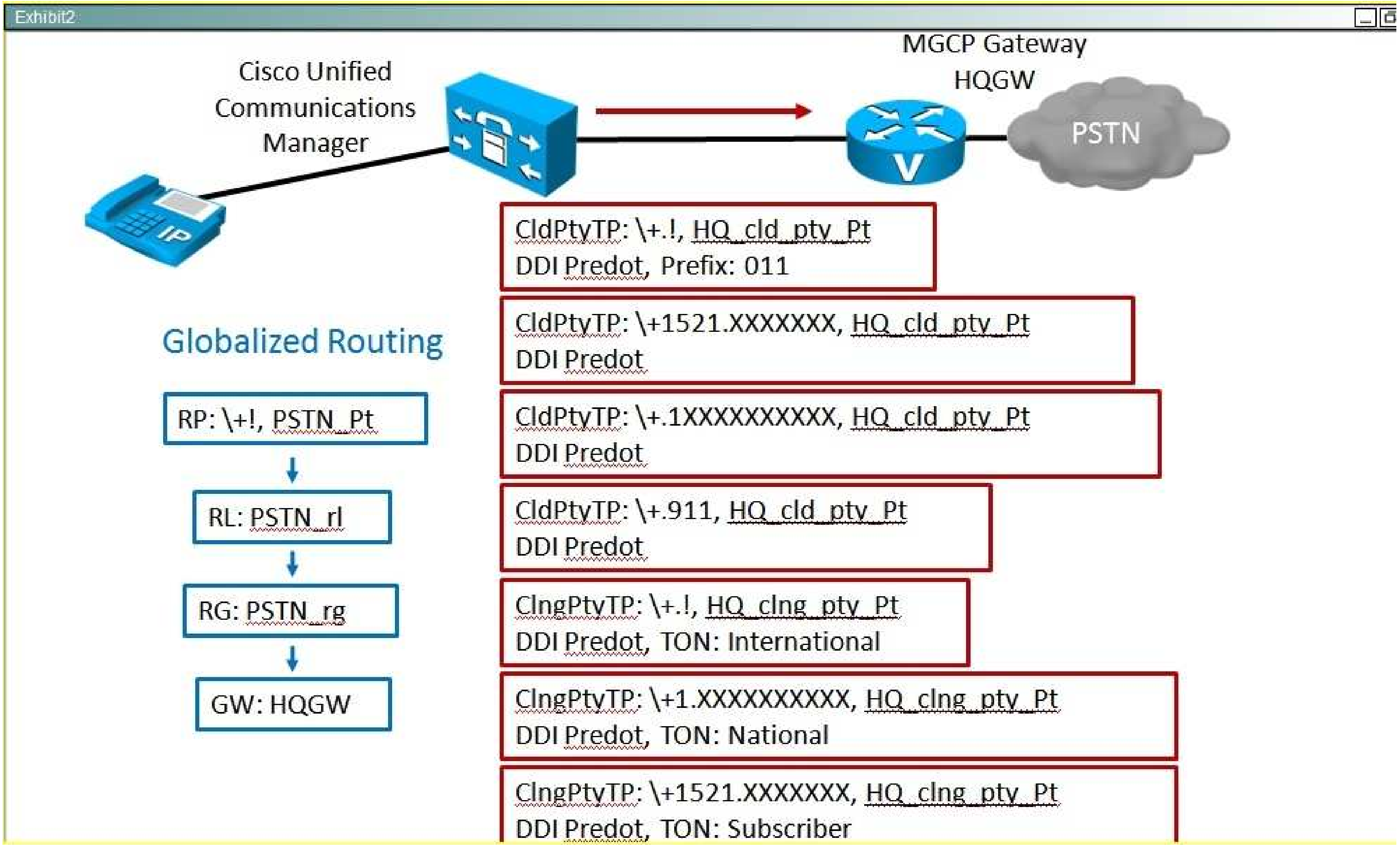


Exhibit1
Cisco Unified Communications Manager
MGCP Gateway
HQGW
PSTN
Globalization at Ingress
DN: 3001, Internal_Pt
Ext. mask: +1521555XXXX
CSS: Manager.css
TP: 9.911, HQ_Emergency_Pt
CSS: PSTN.css
DDI Predot, Prefix: +
TP: 9.[2-9]XXXXXX, HQ_Local_Pt
CSS: PSTN.css
DDI Predot, Prefix: +1521
TP: 91.[2-9]XX[2-9]XXXXXX, HQ_LD_Pt
CSS: PSTN.css
DDI Predot, Prefix: +1
TP: 9011.!#, HQ_Intl_Pt
CSS: PSTN.css
DDI Predot Trailing - #, Prefix: +
TP: \+!, Internal_Pt
CSS: PSTN.css
Manager.css:
HQ_Emergency_Pt,
HQ_Local_Pt,
HQ_LD_Pt,
HQ_Intl_Pt
PSTN.css:
PSTN_Pt
Internal.css:
Internal_Pt



The MGCP gateway has the following configurations:

called party transformation CSS HQ_cld_pty CSS (partition=HQ cld_pty.Pt) call.ng party transformation CSS HQ_clng_pty CSS (partition=HQ_clng_pty Pt)

All translation patterns have the check box "Use Calling Party's External Phone Number Mask" enabled.

When the IP phone at extension 3001 places a call to 9011 49403021 56001# what is the resulting called and calling number that is sent to the PSTN?

A. The called number is 01 1 49403021 56001. The calling number will be 5553001 and number type set to subscriber.
B. The called number is 011 49403021 56001. The calling number will be 5215553001 and number type set to national.
C. The called number is 4940302156001 with number type set to international. The calling number will be 5215553001 and number type set to national.
D. The called number is +49403021 56001 with number type set to international. The calling number will be 5215553001 and number type set to subscriber.

Correct Answer: A
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:
Incorrect answer: B, C, D
Explanation: Check the check box "Use Calling Party's External Phone Number Mask" if you want the full, external phone number to be used for calling line identification (CLID) on outgoing calls. You may also configure an External Phone Number Mask on all phone devices.
Link: http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_tech_note09186a00805b6f33.shtml

QUESTION 63
Refer to the exhibit

When the Cisco Unified Communications Manager advertises the Hosted DN Pattern, which pattern would be advertised?

A. 2XXX and the T0D1D will be 0:+498950555
B. 2XXX and the ToDID will be 0:+4989531 21
C. 4989S05552XXX and the ToDiD will be 0:
D. + 4989631 21 2XXX and the ToDiD will be 0:
E. Both +4989505552XXXand +4989531 21 2XXX will be advertised with ToDID of 0:

Correct Answer: A
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:
Incorrect answer: B, C, D, E
Explanation: PSTN failover prepend digit is +498950555
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_0_2/ccmfeat/fscallcontroldiscovery.html

QUESTION 64
When an external call is placed from Ajax, they would like the ANI that is sent to the PSTN to be the main number, not the extension. For domestic calls, they would like 10 digits sent; for international calls, they would like to send the country code 1 and the 10 digits. How can this be accomplished?

A. Add a translation pattern to the dial peers in the gateway that adds the appropriate digits to the outgoing ANI.
B. In the external call route patterns, set the external phone number mask to the main number. Use 10 digits in the domestic route pattern and 1 followed by the main number digits in the international route patterns.
C. Use a calling party transform mask for each route group in the corresponding route list configuration. Set the explicit 10-digit main number for domestic calls and 1 followed by the main number for the international route patterns.

D. In the directory number configurations, set the prefix digits field to the country code and the 10 digits of the main number. This will be truncated to the 10-digit number for domestic calls and sent out in its entirety for international calls.

Correct Answer: C
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:
Incorrect answer: A, B, D
Explanation: calling party transformation mask value is Valid entries for the NANP include the digits 0 through 9; the wildcard characters X, asterisk (*), and octothorpe (#); and the international escape character +.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b03trpat.html

QUESTION 65
Which trunk should you use in an H.323 gatekeeper-controlled network?

A. H.323
B. H.225
C. SIP
D. Intercluster
E. MGCP FXO trunk
F. MGCP T1/E1 trunk

Correct Answer: B
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 66
Which sign is prefixed to the number in global call routing?

A. -
B. +
C. #
D. @
E. &

F. *

Correct Answer: B
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 67
Which device is used to connect to the H.323 gatekeeper?

A. H.323 gateway
B. SIP trunk
C. H.323 trunk
D. MGCP gateway

Correct Answer: C
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 68
The corporate WAN has been extended to two newly acquired sites and it includes gatekeeper support. Each site has a Cisco CallManager and an H.323 gateway that allows connection to the PSTN. Which connection method is best for these two new customers?

A. H.225 trunk (gatekeeper-controlled)
B. intercluster trunk (non-gatekeeper controlled)
C. SIP trunk
D. intercluster trunk (gatekeeper-controlled)

Correct Answer: D
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 69
Which E.164 transformation pattern represents phone numbers in Germany?

A. \+49.!
B. 49.!
C. \49.!
D. \+49.X

Correct Answer: A
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

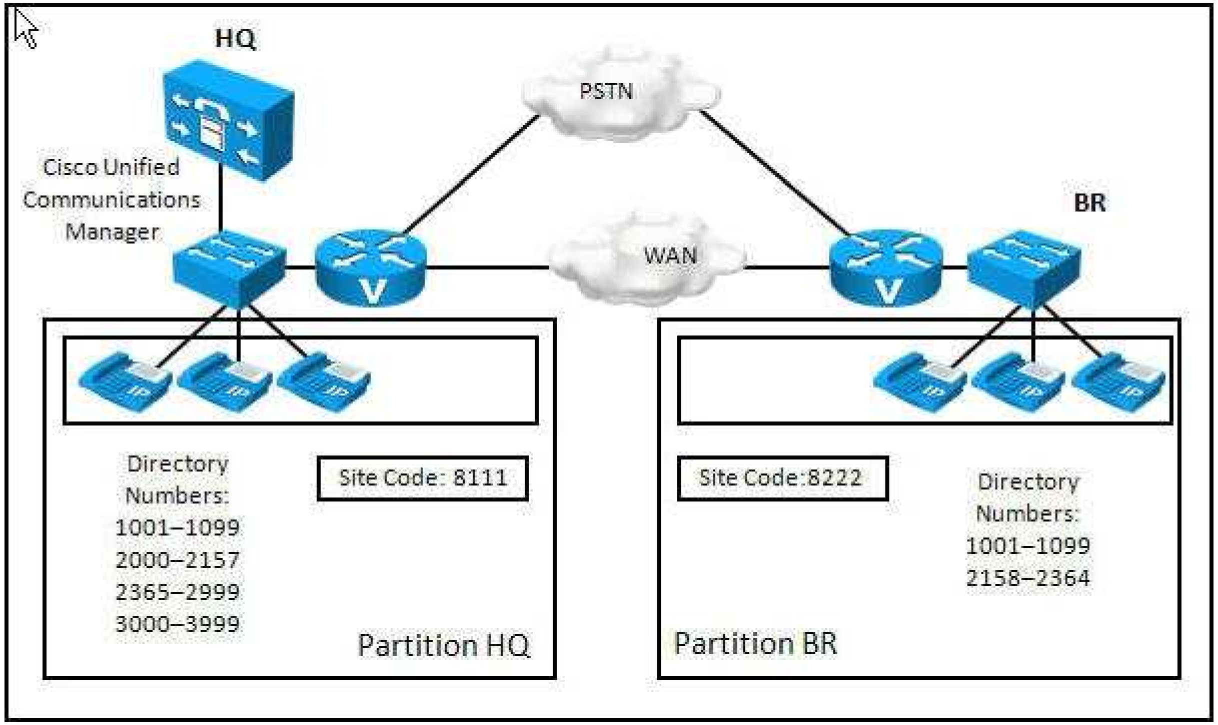
QUESTION 70
Which two statements are true regarding the implementation of globalized call-routing in terms of localized call egress? (Choose two.)

A. Calling-party numbers are routed from the gateway or trunks to phones.
B. Called-party numbers are routed from the gateway or trunks to phones.
C. Calling-party numbers of internal calls are routed from the gateway or trunks.
D. Calling-party calls are routed to the gateway and trunks.

Correct Answer: AD
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 71
Refer to the exhibit. Assume that the HQ phones have access to the HQ partition, and BR phones have access to the BR partition. Which set of implementations would best address the overlapping directory number extensions for intersite (WAN) calling between the HQ site and the BR site?



A. Configure a route pattern 8222.[12]XXX for site HQ, and assign it to partition HQ. Configure the called party DDI of Predot.
Configure a route pattern for site BR 8111.[1-3]XXX, and assign it to partition BR. Configure called party DDI Predot.
Use the local gateway at each site. Prefix the appropriate site code for the calling number.

B. Configure a single route pattern for both sites 8[12,12,12].[1-32]XXX. Use a route list that contains the local route group for each site. Prefix the appropriate site code for the calling number.

C. Configure a translation pattern 8222.[12]XXX for site HQ, and assign it to partition HQ. Use a CSS that contains the partitions for BR phones.
Configure a translation pattern 8111.[1-3]XXX for site BR, and assign it to partition BR. Use a CSS that contains the partitions for HQ phones.
For both translation patterns, configure the called party DDI of Predot. Prefix the appropriate site code for the calling number.

D. Configure a translation pattern 8222.[12]XXX for site HQ, and assign it to partition BR. Use a CSS that contains the partitions for HQ phones.
Configure a translation pattern 8111.[1-3]XXX for site BR, and assign it to partition HQ. Use a CSS that contains the partitions for BR phones.
For both translation patterns, configure the called party DDI of Predot. Prefix the appropriate site code for the calling number.

Correct Answer: C
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 72
What is the difference between an H.323 gateway and a SIP gateway?

A. An H.323 gateway requires that dial peers be configured before PSTN calls can be placed and received. The SIP gateway requires no dial peers.
B. The H.323 gateway can be added in Cisco Unified Communications Manager under gateway type as H.323 Gateway. The SIP gateway can connect to Cisco Unified Communications Manager only through a SIP trunk.
C. A SIP gateway requires a call agent for PSTN calls to be placed and received. An H.323 gateway does not require a call agent for PSTN calls to be placed and received.
D. An H.323 gateway can register with Cisco Unified Communications Manager. A SIP gateway will show status of "Unknown".
E. The H.323 gateway must be configured in Cisco Unified Communications Manager using a valid IP address on the gateway. The SIP gateway must be configured in Cisco Unified Communications Manager using the domain name.

Correct Answer: B
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 73
What is the difference between an MGCP gateway and a SIP gateway?

A. An MGCP gateway that dial peers be configured before PSTN calls can be placed and received. The SIP gateway requires no dial peers.
B. An MGCP gateway can be added in Cisco Unified Communications Manager under the Gateway Type field using the gateway model. The SIP gateway can connect to Cisco Unified Communications Manager only through a SIP trunk.
C. A SIP gateway requires a call agent for PSTN calls to be placed and received. An MGCP gateway does not require a call agent for PSTN calls to be placed and received.
D. An MGCP gateway can register with Cisco Unified Communications Manager. A SIP gateway will show status of "Unknown".
E. The SIP gateway must be configured in Cisco Unified Communications Manager using a valid IP address on the gateway. The MGCP gateway must be configured in Cisco Unified Communications Manager using the domain name.

Correct Answer: B
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

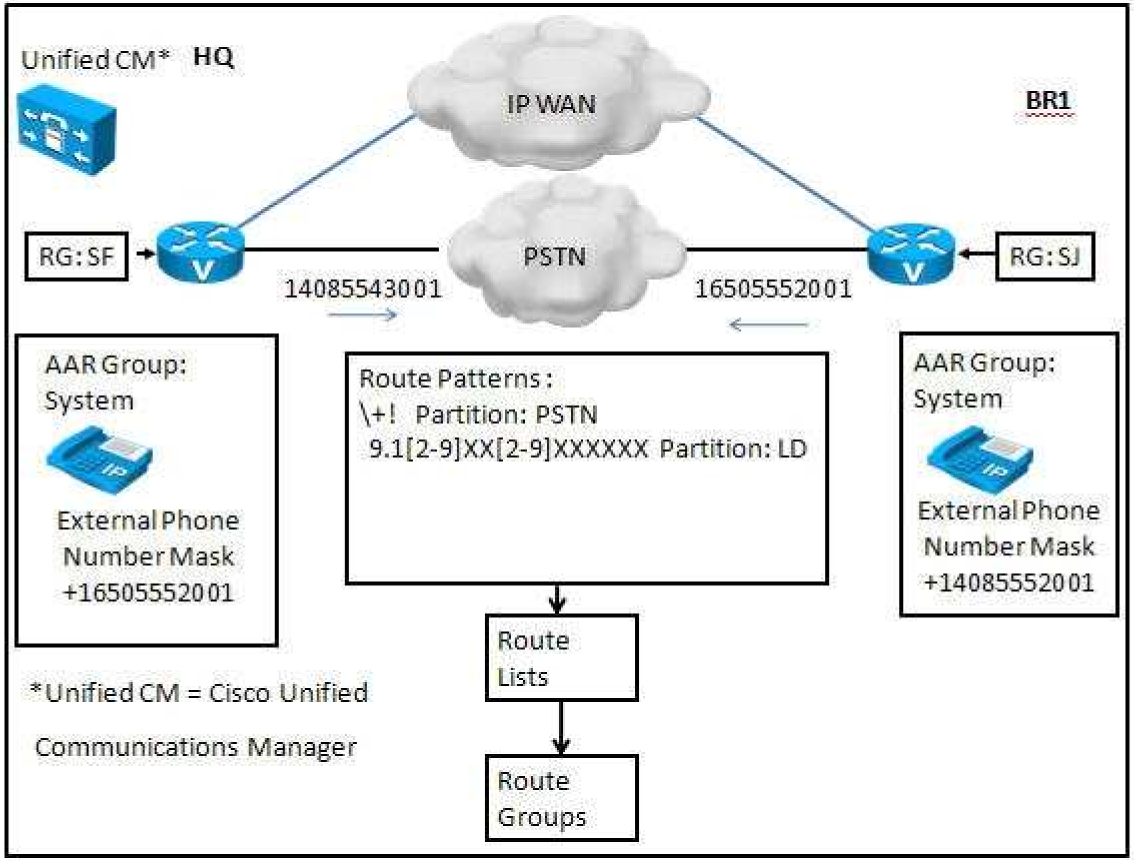
QUESTION 74
When an incoming PSTN call arrives at an H.323 gateway, how does the calling number get normalized to a global E.164 number with + prefix in Cisco Unified Communications Manager?

A. Normalization is done using translation patterns.
B. Normalization is done using route patterns.
C. Normalization is done using the gateway incoming called party prefixes based on number type.
D. Normalization is done using the gateway incoming calling party prefixes based on number type.
E. Normalization is achieved by local route group that is assigned to the H.323 gateway.

Correct Answer: D
Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 75
Refer to the exhibit.



The HQ site uses area code 650. The BR1 site uses area code 408. The long distance national code for PSTN dialing is 1. To make a long distance national call, an HQ or BR1 user dials access code 9, followed by 1, and then the 10-digit number. Both sites use MGCP gateways. AAR must use globalized call routing using a single route pattern. Assume that all outgoing PSTN numbers are localized at the egress gateway as shown in the exhibit. Which statement is true?

A. The AAR group system must be configured on the device configuration of the phones.
B. The AAR group system must be configured on the line configuration of the phones.
C. The single AAR group system cannot be used. A second AAR group must be configured in order to have source and destination AAR groups.
D. The AAR group system must be configured under the AAR service parameters.

Correct Answer: B

Section: Describe and Configure a Multi-site Dial Plan for Cisco Unified Communications Manager
Explanation

Explanation/Reference:

QUESTION 76
Refer to the exhibit.

```
router eigrp SAF
 !
 service-family ipv4 automous-system 1
  !
   sf-interface FastEthernet0/0
   topology base
    exit-sf-topology
 exit-service-family
```

Which configuration elements must match for adjacent neighbors to establish a SAF neighbor relationship?

A. the label name specified in the router eigrp command
B. the autonomous-system number specified in the service-family ipv4 autonomous-system command
C. the sf-interface configuration
D. the topology base configurations
E. the label name specified in the router eigrp command and the autonomous-system number

Correct Answer: B
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Incorrect answer: A, C, D, E
Explanation: service-family ipv4 autonomous-system 1 enables a Cisco SAF service family for the specified autonomous system on the router
Link: http://www.cisco.com/en/US/docs/ios/saf/configuration/guide/saf_cg_ps10591_TSD_Products_Configuration_Guide_Chapter.html#wp1056363

QUESTION 77
Which statement about Service Advertisement Framework is true?

A. SAF requires that the EIGRP be configured on all routers, including non-SAF routers.
B. SAF requires that the EIGRP be configured only on SAF routers. Non-SAF routers act as an IP cloud.
C. SAF has no dependency on the underlying routing protocol, as long as it is a dynamic routing protocol. Static routes are not supported.
D. SAF operates on any dynamic or static IP routing configuration. SAF is totally independent of the underlying routing protocol.

Correct Answer: D
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Explanation:
Because Cisco SAF is independent of IP routing and uses underlying Cisco routing technology to distribute service advertisements in a reliable and efficient manner, Cisco SAF will run in networks over any routing protocol they may have in place such as Enhanced Interior Gateway Routing Protocol (EIGRP), Open Shortest Path First (OSPF), Exterior Border Gateway Protocol (EBGP) over an MPLS service, or static routing (Figure 2).

http://www.cisco.com/en/US/prod/collateral/iosswrel/ps6537/ps6554/ps6599/ps10822/whitepaper_c11-636604.html

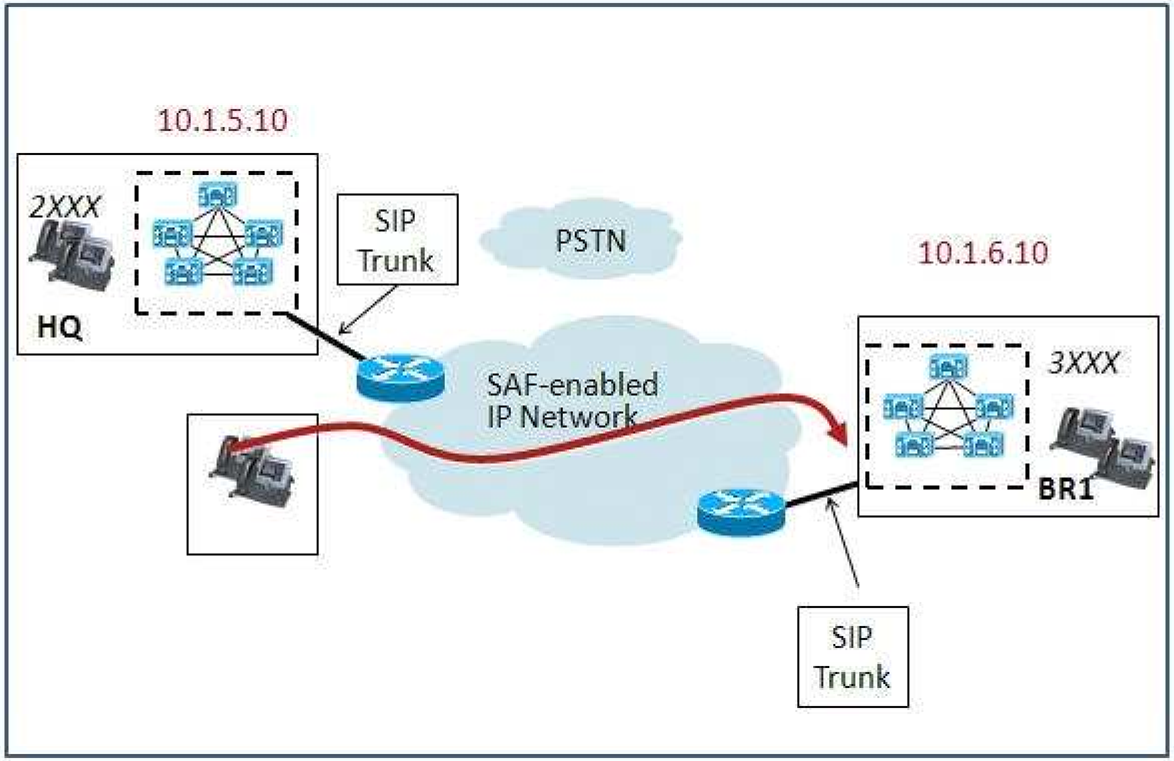
QUESTION 78
Assume that the Cisco IOS SAF Forwarder is configured correctly. Which minimum configurations on Cisco Unified Communications Manager are needed for the SAF registration to take place?

A. SAF Trunk, SAF Security Profile, SAF Forwarder, and CCD Advertising Service
B. SAF Trunk, SAF Security Profile, SAF Forwarder, and CCD Requesting Service
C. SAF Trunk, SAF Security Profile, SAF Forwarder, CCD Requesting Service, and CCD Advertising Service
D. SAF Trunk, SAF Security Profile, and SAF Forwarder
E. SAF Trunk, CCD Requesting Service, and CCD Advertising Service

Correct Answer: B
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 79
Refer to the exhibit.



What should the destination IP address be configured as on the HQ and BR1 SIP trunks?

A. The HQ SIP trunk destination IP address should be 10.1.6.10. The BR1 SIP trunk destination IP address should be 10.1.5.10.
B. The destination IP address is not configurable. Each cluster will learn about the remote trunk IP address through SAF learned routes.
C. The destination IP address will be learned automatically and configured on the SIP trunks after the Cisco Unified Communications Managers discover themselves.
D. The HQ SIP trunk destination IP address should be the HQ SAF Forwarder IP address. The BR1 SIP trunk destination IP address should be the BR1 SAF Forwarder IP address.

Correct Answer: B
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Incorrect answer: A, C, D
Explanation: The gatekeeper changes the IP address of this remote device dynamically to reflect the IP address of the remote device.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a08trnk.html

QUESTION 80
When a SIP trunk is added for Call Control Discovery, which statement is true?

A. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Enable SAF check box should be selected.
B. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The Trunk Service Type should be Call Control Discovery.
C. The SIP trunk is added by selecting Call Control Discovery Trunk and then selecting SIP as the protocol to be used.
D. The SIP trunk is added by selecting SIP Trunk and SIP Protocol. The destination IP address field is configured as ‘SAF’ to indicate that this trunk is used for SAF.

Correct Answer: B
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 81
When an H.323 trunk is added for Call Control Discovery, which statement is true?

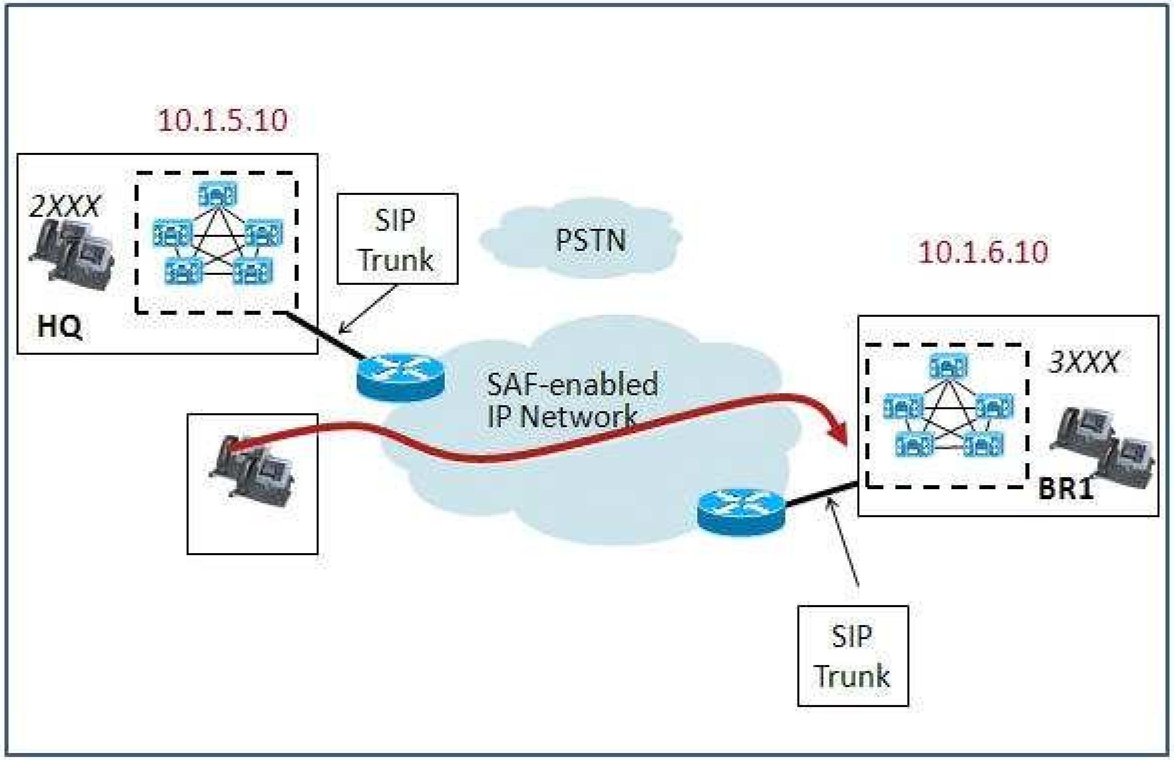
A. The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Enable SAF check box should be selected in the trunk configuration.
B. The H.323 trunk is added by selecting Inter-Cluster Trunk (Non-Gatekeeper Controlled) and Device Protocol Inter-Cluster Trunk. The Trunk Service Type should be Call Control Discovery.
C. The H.323 trunk is added by selecting Call Control Discovery Trunk and then selecting H.323 as the protocol to be used.
D. The H.323 trunk is added by selecting H.323 Trunk, and selecting Inter-Cluster Trunk as the Device Protocol. The destination IP address field is configured as ‘SAF’ to indicate that this trunk is used for SAF.

Correct Answer: A
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Reference. Implementing Cisco Unified Communications Manager Part 2 (CIPT2), Chapter3: Implementing Multisite Connections, pg 70-71, Fig 3-14 and Fig 3-15

QUESTION 82

Refer to the exhibit.



What must be configured on the HQ Cisco Unified Communications Manager to allow HQ users to dial the SAF learned directory number pattern 3XXX?

A. Route pattern 3XXX should be configured and made available to HQ users through the phone CSS.
B. Route pattern 3XXX should be configured and made available to HQ phone users through the phone AAR CSS.
C. The SAF partition assigned to the SAF learned patterns must be available to the HQ phone users through the phone CSS.
D. The SAF partition assigned to the SAF learned patterns must be available to the HQ phone users through the phone AAR CSS.
E. The SAF directory number pattern 3XXX will be made available to all users automatically as soon as the SAF partition is selected.

Correct Answer: C
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Incorrect answer: A, B, D
Explanation: By adopting the SAF network service, the call control discovery feature allows Cisco Unified Communications Manager to advertise itself along with other key attributes.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fscallcontroldiscovery.html

QUESTION 83
Refer to the exhibit.

```
dial-peer hunt 2
voice service saf
 profile trunk-route 1
  session protocol sip interface Loopback1 transport tcp port 5060
 !
 profile dn-block 1 alias-prefix 1972555
  pattern 1 type extension 4XXX
 !
 profile dn-block 2
  pattern 1 type global 14087071222
 !
 profile callcontrol 1
  dn-service
   trunk-route 1
   dn-block 1
   dn-block 2
  !
 !
 !
 channel 1 vrouter SAF asystem 1
  subscribe callcontrol wildcarded
  publish callcontrol 1
 !
```

Which CSS is used at the HQ Cisco Unified Communications Manager to reroute calls via the PSTN when the SAF network is unavailable?

A. the phone device CSS
B. the phone line CSS
C. the phone line/device combined CSS
D. the SAF CSS configured on the CCD requesting service
E. the phone AAR CSS configured at the phone device
F. No special CSS is required. If SAF patterns are accessible, the PSTN reroute is automatic.

Correct Answer: E
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 84
Refer to the exhibit.

```
dial-peer hunt 2
voice service saf
 profile trunk-route 1
  session protocol sip interface Loopback1 transport tcp port 5060
 !
 profile dn-block 1 alias-prefix 1972555
  pattern 1 type extension 4XXX
 !
 profile dn-block 2
  pattern 1 type global 14087071222
 !
 profile callcontrol 1
  dn-service
   trunk-route 1
   dn-block 1
   dn-block 2
  !
 !
 !
 channel 1 vrouter SAF asystem 1
  subscribe callcontrol wildcarded
  publish callcontrol 1
 !
```

How does the Cisco Unified Communications Manager advertise dn-block 1?

A. 4XXX and the ToDID will 0:
B. 4XXX and the ToDID will 0:1972555
C. 4XXX
D. 4XXX and the ToDID will 0:+ 1972555
E. 19725554XXX

Correct Answer: B
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 85
Refer to the exhibit.

```
dial-peer hunt 2
voice service saf
 profile trunk-route 1
  session protocol sip interface Loopback1 transport tcp port 5060
  !
 profile dn-block 1 alias-prefix 1972555
  pattern 1 type extension 4XXX
  !
 profile dn-block 2
  pattern 1 type global 14087071222
  !
 profile callcontrol 1
  dn-service
   trunk-route 1
   dn-block 1
   dn-block 2
   !
  !
  !
 channel 1 vrouter SAF asystem 1
  subscribe callcontrol wildcarded
  publish callcontrol 1
  !
```

How does the Cisco Unified Communications Manager advertise dn-block 2?

A. 14087071222 with number type international
B. +14087071222 with number type international
C. +14087071222

D. 14087071222

Correct Answer: C
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 86
Which Cisco IOS command is used to verify that a SAF Forwarder that is registered with Cisco Unified Communications Manager has established neighbor relations with an adjacent SAF Forwarder?

A. show eigrp service-family ipv4 neighbors
B. show eigrp address-family ipv4 neighbors
C. show voice saf dndball
D. show saf neighbors
E. show ip saf neighbors

Correct Answer: A
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Incorrect answer: B, C, D, E
Explanation:
Router# show eigrp service-family ipv4 4453 neighbors

EIGRP SFv4 VR(test) Service-Family Neighbors for AS(4453)

Address	Interfaces	Hold	Uptime	SRTT (sec)	RTO ((sec))	Q	Seq (msec)
10.1.1.1	Ethernet0/0	13	00:00:41	30	1014	0	16
10.1.2.1	Ethernet0/0	14	00:02:02	20	200	0	10

Link: http://www.cisco.com/en/US/docs/ios/saf/command/reference/saf_s1.html

QUESTION 87

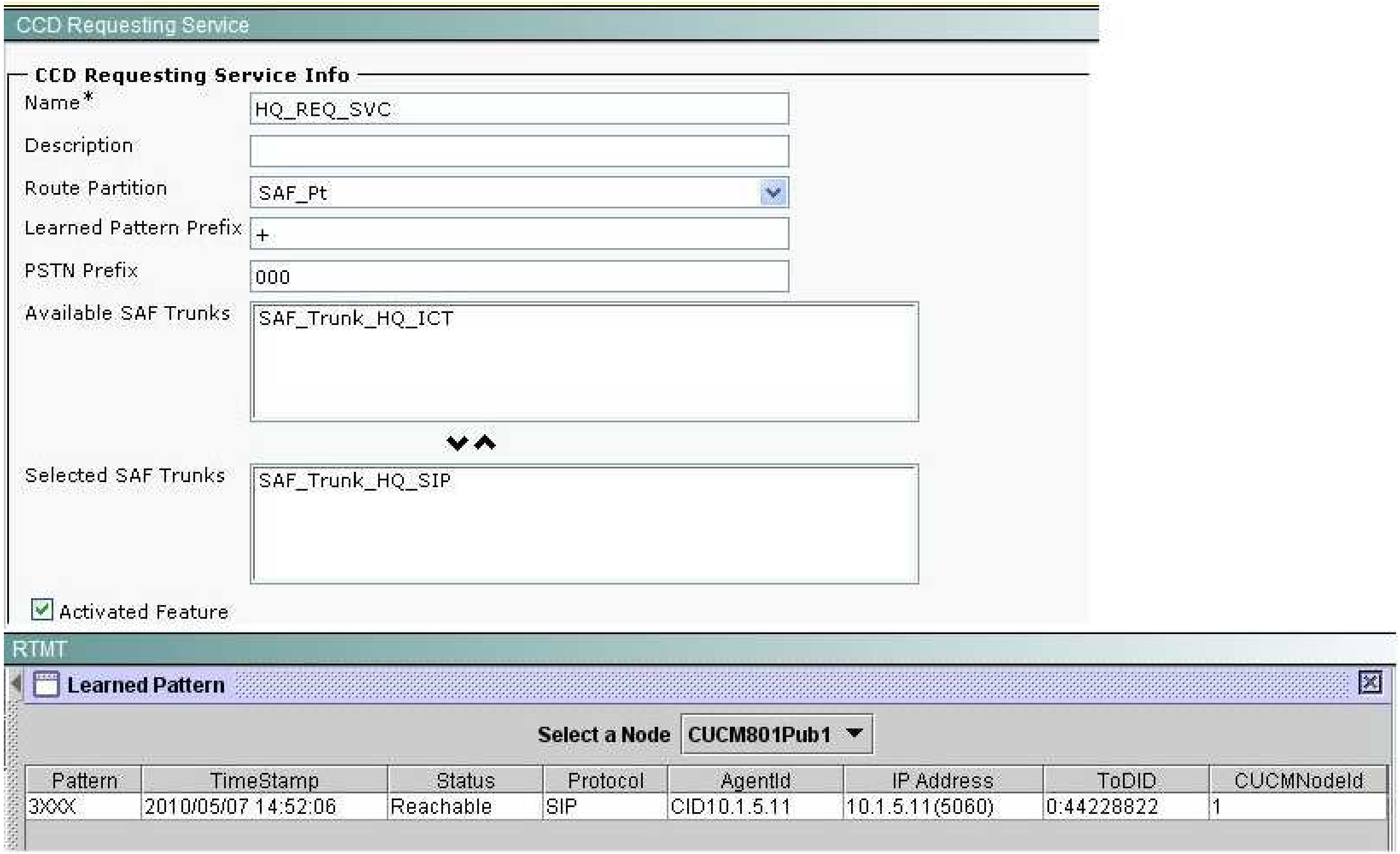
Which Cisco IOS command is used to verify that the Cisco Unified Communications Manager Express has registered with the SAF Forwarder?

A. show eigrp service-family ipv4 clients
B. show eigrp address-family ipv4 clients
C. show voice saf dndb all
D. show saf registration
E. show ip saf registration

Correct Answer: A
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Incorrect answer: B, C, D, E
Explanation: show eigrp service-family ipv4 clients Displays information from the EIGRP IPv4 service-family results.

QUESTION 88
Refer to the exhibit.



When the user of a phone registered to the Cisco Unified Communications Manager places a call to 3001 when the SAF network is down, what happens?

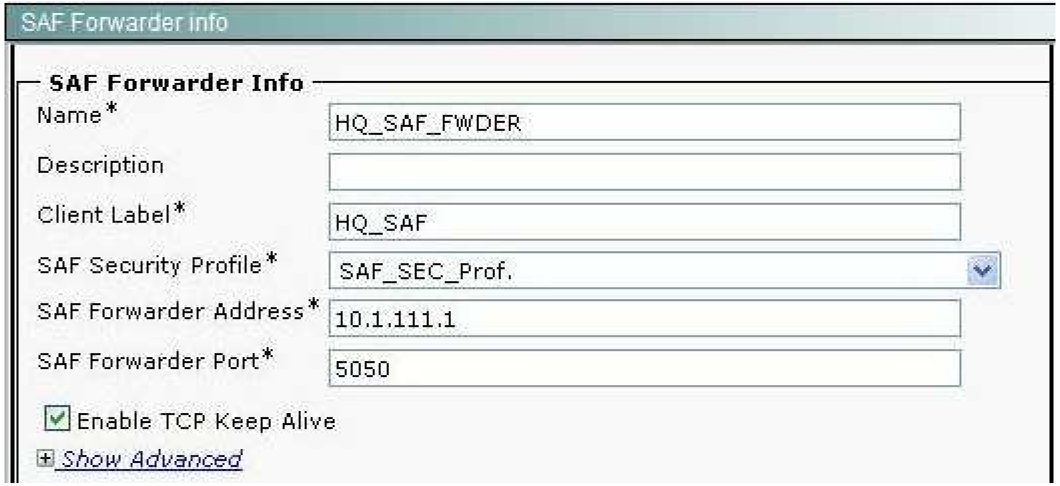
A. The call fails.
B. The call is rerouted to the PSTN with the constructed PSTN number as +442288223001
C. The call is rerouted to the PSTN with the constructed PSTN number as 442288223001
D. The call is rerouted to the PSTN with the constructed PSTN number as 0002288223001

E. The call is rerouted to the PSTN with the constructed PSTN number as +0002288223001

Correct Answer: A
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Incorrect answer: B, C, D
Explanation: When the SAF forwarder loses network connection with its call-control entity, the SAF forwarder withdraws those learned patterns that were published by the call control entity. In this case, CCD requesting service marks those learned patterns as unreachable via IP, and the calls get routed through the PSTN gateway.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fscallcontroldiscovery.html

QUESTION 89
Refer to the following exhibit.



Which Cisco IOS SAF Forwarder configuration is correct?

A

```
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
  topology base
   external-client HQ_SAF
  exit-sf-topology
 exit-service-family
!
!
service-family external-client listen ipv4 5050
 external-client HQ_SAF
  username SAFUSER
  password SAFPASSWORD
  keepalive 3600000
```

B

```
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
  topology base
   external-client HQ_SAF_FWDER
  exit-sf-topology
 exit-service-family
!
!
service-family external-client listen ipv4 5050
 external-client HQ_SAF
  username SAFUSER
  password SAFPASSWORD
  keepalive 3600000
```

```
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
```

A. Exhibit A
B. Exhibit B
C. Exhibit C
D. Exhibit D

Correct Answer: A
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Incorrect answer: B, C, D
Explanation: Summary steps to configure IOS SAF forwarder is given below
1. enable
2. configure terminal
3. router eigrp *virtual-instance-name*
4. service-family {ipv4 **|** ipv6} [vrf *vrf-name*] autonomous-system *autonomous-system-number*
5. topology base
6. external-client *client_label*
7. exit-sf-topology
8. exit-service-family
9. exit
10. service-family external-client listen {ipv4 **|** ipv6} *tcp_port_number*
11. external-client *client-label*
12. username *user-name*
13. password *password-name*
14. keepalive *number*
15. exit

Link: http://www.cisco.com/en/US/docs/ios/saf/configuration/guide/saf_cg_ps11280_TSD_Products_Configuration_Guide_Chapter.html

QUESTION 90
Refer to the exhibit:

IOS SAF Forwarder Config

```
router eigrp SAF
 !
 service-family ipv4 autonomous-system 1
  !
  topology base
   external-client HQ_SAF
  exit-sf-topology
 exit-service-family
!
!
service-family external-client listen ipv4 5050
 external-client HQ_SAF
  username SAFUSER
  password SAFPASSWORD
  keepalive 3600000
```

The exhibit shows a SAF Forwarder configuration attached to a Cisco Unified Communications Manager.
Which minimum configuration for a Cisco Unified Communications Manager Express SAF Forwarder is needed to establish a SAF neighbor relationship with this SAF Forwarder?

A. router eigrp SAF
i
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
voice service saf
profile trunkroute 1
session protocol sip interface Loopback1 transport tcp port 5060
!

B. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit- service-family

```
!
voice service saf
profile trunk-route 1
session protocol sip interface Loopback1 transport tcp port 5060
!
profile dn-block 1 alias-prefix 1972555
pattern 1 type extension 4xxx
!
 profile callcontrol 1
dn-service
trunk-route 1
dn-block 1
dn-block 2
!
channel 1 vrouter SAF asystem 1
subscribe callcontrol wildcarded
publish callcontrol 1
!
```

C. router eigrp SAF
```
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
```

D. None of above configurations contain sufficient information.

Correct Answer: C
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Incorrect answer: A, B, D
Explanation: only following configuration is enough
router eigrp SAF
service-family ipv4 autonomous-system 1
exit-service-family
link: http://www.cisco.com/en/US/prod/collateral/iosswrel/ps6537/ps6554/ps6599/ps10822/whitepaper_c11-636604.html

QUESTION 91

Which Cisco IOS command is used for internal SAF Clients to check SAF learned routes?

A. show eigrp address-family ipv4 saf
B. show voice saf routes
C. show voice saf detail
D. show eigrp service-family ipv4 saf
E. show voice saf dndb all

Correct Answer: E
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:
Incorrect answer: A, B, C, D
Explanation: Router# show voice saf dnDb all
Total no. of patterns in db/max allowed : 1/6000
Patterns classified under dialplans (private/global) : 0/1
Informational/Error stats -
Patterns w/ invalid expr detected while add : 0
Patterns duplicated under the same instance : 0
Patterns rejected overall due to max capacity : 0
Attempts to delete a pattern which is invalid : 0

Last successful DB update @ 2009:12:14 15:42:45:967
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/feature/guide/SAF_FeatureModule.html#wp1201815

QUESTION 92
What are the two tasks that you must perform to configure the Service Advertisement Framework forwarder in Cisco Unified Communications Manager? (Choose two.)

A. create VPN groups
B. create VPN profiles
C. create a new Service Advertisement Framework security profile
D. set feature configuration parameters of Call Control Discovery
E. configure Service Advertisement Framework forwarder information
F. enable enterprise parameter for Service Advertisement Framework forwarder

Correct Answer: CE
Section: Implement Call Control Discovery/ILS

Explanation

Explanation/Reference:

QUESTION 93
You are entering the description for the Service Advertisement Framework forwarder. Which three characters should you avoid entering in the description? (Choose three.)

A. < >
B. &
C. %
D. #
E. $
F. @

Correct Answer: ABC
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 94
In a cluster-wide deployment, what is the maximum number of Service Advertisement Framework forwarders to which the Cisco Unified Communications Manager can connect?

A. 1
B. 2
C. 3
D. 4
E. 6
F. as many as are configured

Correct Answer: B
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 95
In a node-specific Service Advertisement Framework forwarder deployment model, what is the maximum number of Service Advertisement Framework forwarders that you can assign to a specific node?

A. 1
B. 2
C. 3
D. 4
E. 5
F. 6

Correct Answer: B
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 96
Which component of Cisco Unified Communications Manager is responsible for sending keepalive messages to the Service Advertisement Framework forwarder?

A. Call Control Discovery requesting service
B. Hosted DNs service
C. Service Advertisement Framework client control
D. Cisco Unified Communications Manager database
E. Service Advertisement Framework-enabled trunk
F. gatekeeper

Correct Answer: C
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 97
Which two actions are performed by the Call Control Discovery service after the local Cisco Unified Communications Manager loses its TCP connection with the

primary and secondary Service Advertisement Framework? (Choose two.)

A. Calls are routed to the PSTN gateway after the Call Control Discovery Learned Pattern IP Reachable Duration parameter expires.
B. All learned patterns are purged from the local cache after the Call Control Discovery PSTN Failover Duration parameter expires.
C. The Service Advertisement Framework forwarder contacts all the remaining Service Advertisement Framework forwarders in the cluster.
D. All the remaining Service Advertisement Framework forwarders are notified for their learned patterns.
E. The Cisco Unified Communications Manager establishes a connection with the primary and secondary Service Advertisement Framework after the Learned Pattern IP Reachable Duration parameter expires.
F. Call Control Discovery immediately redirects all the calls to the PSTN gateway based on the learned patterns.

Correct Answer: AB
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 98
Which minimum configuration is needed for the SAF Internal Client to register with this SAF Forwarder?

A. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
voice service saf
profile trunk-route 1
session protocol sip interface Loopbackl transport tcp port 5060
i
B. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!

voice service saf
profile trunk-route 1
session protocol sip interface Loopbackl transport tcp port 5060
!
profile dn-block 1 alias-prefix 1972555
pattern 1 type extension 4xxx
!
profile callcontrol 1
dn-service
trunk-route 1
dn-block 1
dn-block 2
i

C. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base
exit-sf-topology
exit-service-family
!
voice service saf
profile trunk-route 1
session protocol sip interface Loopbackl transport tcp port 5060
!
profile dn-block 1 alias-prefix 1972555
pattern 1 type extension 4xxx
!
profile callcontrol 1
dn-service
trunk-route 1
dn-block 1
dn-block 2
!
channel 1 vrouter SAF asystem 1
subscribe callcontrol wildcarded
publish callcontrol 1
i

D. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!

topology base
exit-sf-topology
exit-service-family
!
voice service saf
!
channel 1 vrouter SAF asystem 1

E. router eigrp SAF
!
service-family ipv4 autonomous-system 1
!
topology base exit-sf-topology
exit-service-family i

Correct Answer: A
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 99
Which statement is true regarding the configuration of SAF Forwarder?

A. In a multisite dial plan, SAF Forwarders may exist in multiple autonomous systems.
B. The client label that is configured in Cisco Unified Communications Manager must match the configuration on the SAF Forwarder router.
C. There should not be multiple nodes of Cisco Unified Communications Manager clusters acting as SAF clients.
D. The destination IP address must match the loopback address of the SAF router.

Correct Answer: A
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 100
Which three devices support the SAF Call Control Discovery protocol? (Choose three.)

A. Cisco Unified Border Element

B. Cisco Unity Connection
C. Cisco IOS Gatekeeper
D. Cisco Catalyst Switch
E. Cisco IOS Gateway
F. Cisco Unified Communications Manager

Correct Answer: AEF
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 101
Refer to the exhibit. When a user presses a speed dial to +442079460255 when the SAF network is down, which event should occur?

Learned Pattern

Select a Node **CUCM801Pub1**

Pattern	TimeStamp	Status	Protocol	AgentId	IP Address	ToDID
\+4420!	2010/05/05 09:49:03	Reachable	SIP	CID10.1.5.11	10.1.5.11(5060)	0:
\+4420!	2010/05/05 09:49:03	Reachable	H323	CID10.1.5.11	10.1.5.11(47005)	0:

A. The call will reroute via the PSTN with the constructed PSTN number as 442079460255.
B. The call will reroute via the PSTN with the constructed PSTN number as +442079460255.
C. The call will reroute via the PSTN with the constructed PSTN number as 00442079460255.
D. The call will fail because the ToDID is 0:.
E. The call will fail because the called number will be 2079460255.

Correct Answer: B
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 102
What is the purpose of a SAF Client?

A. To decode address information and route calls to and from the end points
B. To pass IP information from the CUCM to the endpoint
C. To learn about and advertise or subscribe information about SAF network services
D. To reside in the Cisco IOS software, and to communicate with the SAF forwarder

Correct Answer: C
Section: Implement Call Control Discovery/ILS
Explanation

Explanation/Reference:

QUESTION 103
Assume that local route groups are configured. When an IP phone moves from one device mobility group to another, which two configuration components are not changed? (Choose two.)

A. IP subnet
B. user settings
C. SRST reference
D. region
E. phone button settings

Correct Answer: BE
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:
Incorrect answer: A, C, D
Explanation: Although the phone may have moved from one subnet to another, the physical location and associated services have not changed.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/7_1_2/ccmfeat/fsdevmob.html#wp1137460

QUESTION 104
Where do you specify the device mobility group and physical location after they have been configured?

A. phones
B. DMI

C. device mobility CSS
D. device pool
E. MRGL
F. locale

Correct Answer: D
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:
Incorrect answer: A, B, C, E
Explanation: Before you configure a device pool, you must configure the following items if you want to choose them for the device pool, Cisco Unified Communications Manager group (required), Date/time group (required). Region (required) , SRST reference (optional). Media resource group list (optional), Calling search space for auto-registration (optional). Reverted call focus priority (optional), Device mobility group (optional), Device mobility calling search space, Physical location (optional). Location, AAR group. AAR calling search space.
Link: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmcfg/b02devpl.html

QUESTION 105
Which two options for a Device Mobility-enabled IP phone are true? (Choose two.)

A. The phone configuration is not modified.
B. The roaming-sensitive parameters of the current (that is, the roaming) device pool are applied.
C. The user-specific settings determine which location-specific settings are downloaded from the Cisco Unified Communications Manager device pool.
D. If the DMGs are the same, the Device Mobility-related settings are also applied.

Correct Answer: BD
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

QUESTION 106
What impact do roaming-sensitive settings and Device Mobility settings have on call routing?

A. Device Mobility settings have no impact on call routing, but roaming-sensitive settings modify the AAR group, AAR CSS, and device CSS.
B. Device Mobility settings modify the device CSS and the roaming-sensitive settings modify the AAR group and AAR CSS.
C. Device Mobility settings modify the AAR group and the AAR CSS, the roaming-sensitive settings modify the device CSS.
D. Roaming-sensitive settings are settings that do not have an impact on call routing. Device Mobility settings, on the other hand, may have an impact on call

routing because they modify the device CSS, AAR group, and AAR CSS.

Correct Answer: D
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

QUESTION 107
When Cisco Extension Mobility is implemented, which CSS is used for calling privileges?

A. The user device profile line CSS combined with the device CSS of the physical phone used to log in the extension mobility user.
B. The user device profile device CSS combined with the line CSS of the physical phone used to log in the extension mobility user.
C. Only the user device profile device CSS is used.
D. The combined line/device CSS of the physical phone is used to log in the extension mobility user.
E. The combined line/device CSS of the user device profile.

Correct Answer: A
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

QUESTION 108
When multiple Cisco Extension Mobility profiles exist, which actions take place when a user tries to log in to Cisco Extension Mobility?

A. The login will fail because only a single Cisco Extension Mobility profile is allowed.
B. The user must select the desired profile.
C. The user must login to both profiles in the order they are presented.
D. The user may login to both profiles in any order.
E. Login will only be allowed to multiple profiles if the service parameter Allow Multiple Logins is enabled.

Correct Answer: B
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

Incorrect answer: A, C, D, E
Explanation: Users access Cisco Extension Mobility by pressing the Services or Applications button on a Cisco Unified IP Phone and then entering login information in the form of a Cisco Unified Communications Manager UserID and a Personal Identification Number (PIN). If a user has more than one user device profile, a prompt displays on the phone and asks the user to choose a device profile for use with Cisco Extension Mobility.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmfeat/fsem.html

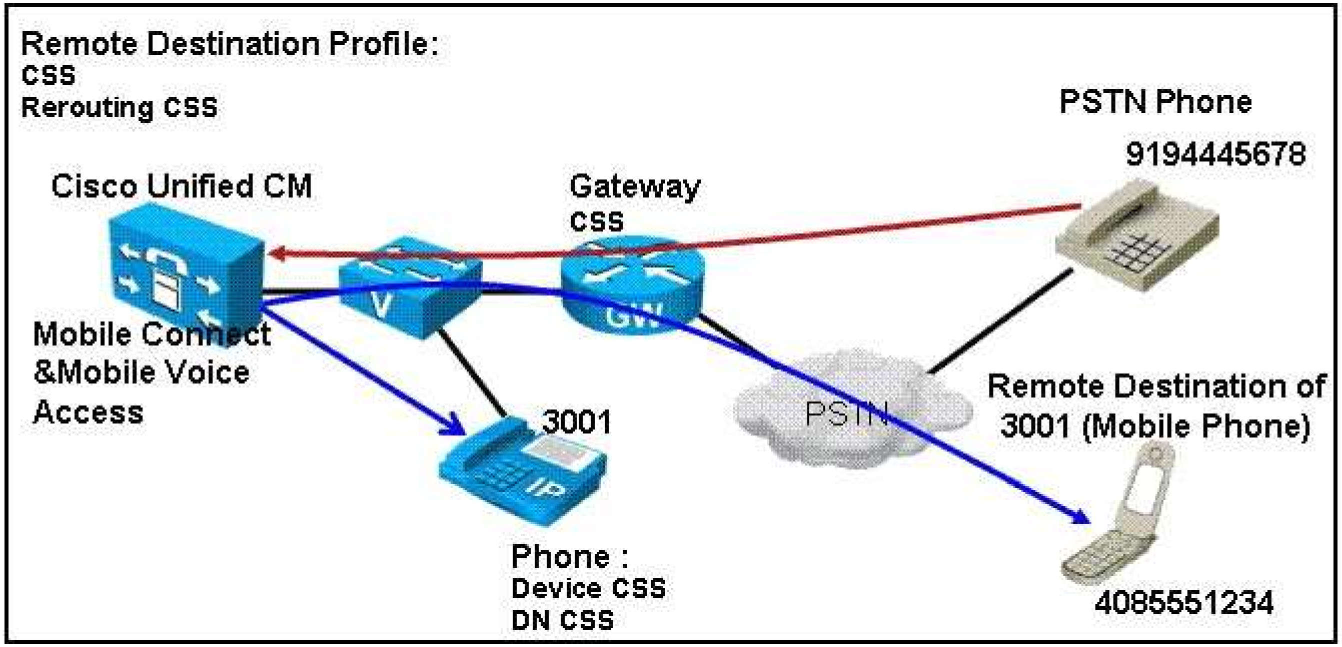
QUESTION 109
When Cisco Extension Mobility is implemented, how is the audio source for the MOH selected?

A. The audio source that is configured at the user device profile is selected.
B. The audio source that is configured at the home phone of the user is selected.
C. The audio source that is configured at the physical phone used for the Cisco Extension Mobility login is selected.
D. The audio source that is configured in the IP Voice Media Streaming parameters is selected.

Correct Answer: A
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:
Incorrect answer: B, C, D
Explanation: To specify the audio source that plays when a user initiates a hold action, choose an audio source from the User Hold MOH Audio Source drop-down list box from device profile configuration settings.
Link: http://cisco.biz/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b06dvprf.html

QUESTION 110
Refer to the exhibit. With the Mobile Connect feature configured, when the PSTN phone calls the Enterprise user at extension 3001, the Enterprise user's mobile phone does not ring. Which CSS is responsible for ensuring that the correct partitions are accessed when calls are sent to the Enterprise user's mobile phone?



A. the gateway CSS
B. the Phone Device CSS
C. the Remote Destination Profile CSS
D. the Remote Destination Profile Rerouting CSS
E. the Phone Line (DN)CSS

Correct Answer: D
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:
Incorrect answer: A, B, C, E
Explanation: Ensure that the gateway that is configured for routing mobile calls is assigned to the partition that belongs to the Rerouting Calling Search Space. Cisco Unified Communications Manager determines how to route calls based on the remote destination number and the Rerouting Calling Search Space.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmfeat/fsmobmgr.html

QUESTION 111
You have been asked to deploy Cisco Extension Mobility Cross Cluster for a distributed call processing environment. During the initial extension mobility login request, how does the visiting cluster determine if the user is a local user or a remote user?

A. by using a third-party automatic provisioning tool to verify user ID
B. by broadcasting a request to all clusters to verify the user type
C. from user IDs that are created by default when the user logs in
D. by using Extension Mobility Cross Cluster Session Initiation Protocol (SIP) trunks
E. by verifying against the local database
F. by verifying the visiting Trivial File Transfer Protocol

Correct Answer: E
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

QUESTION 112
When device mobility mode is enabled or disabled for a cluster, to which does the cluster setting apply?

A. all phones in the cluster that support device mobility
B. all phones in the cluster that subscribed to device mobility
C. mobile phones in the cluster that support device mobility
D. mobile phones in the cluster that are in default mode

Correct Answer: A
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

QUESTION 113
Which statement is true when device mobility mode is enabled or disabled in the Phone Configuration window?

A. The device mobility mode phone settings take precedence over the service parameter settings.
B. The service parameter settings take precedence over the device mobility mode phone settings.

C. The combined service parameter settings and the device mobility mode phone settings will be used.
D. The default settings will be used due to the conflicts.

Correct Answer: A
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

QUESTION 114
Which two entities could be represented by device mobility groups? (Choose two.)

A. countries
B. regions
C. directory numbers
D. transcoders

Correct Answer: AB
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

QUESTION 115
With Cisco Extension Mobility, when a user logs in to a phone type which has no user device profile, what will happen to the phone?

A. The phone takes on the default clusterwide device profile.
B. The phone creates a new device profile automatically.
C. The phone immediately logs the user off.

D. The phone crashes and reboots.

Correct Answer: A
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

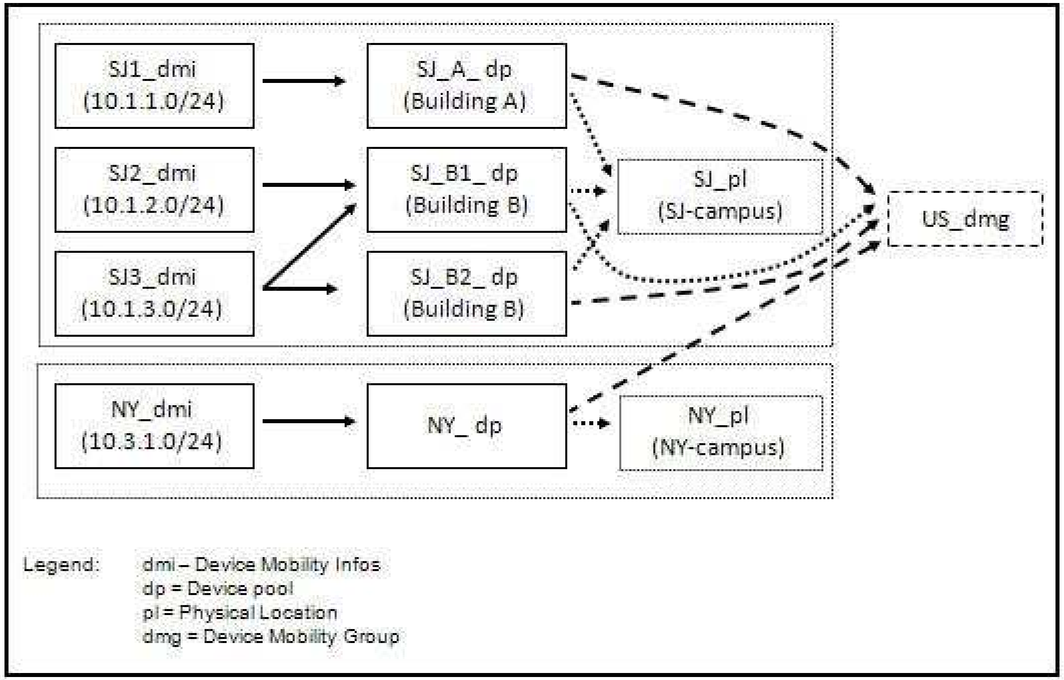
QUESTION 116
Which two Cisco Extension Mobility attributes are available in the user device profile? (Choose two.)

A. regions
B. description
C. phone button template
D. NTP information

Correct Answer: BC
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

QUESTION 117
Refer to the exhibit.
If an IP phone in San Jose roams to New York, which two IP phone settings will be modified by Device Mobility so that the phone can place and receive calls in New York? (Choose two.)



A. The physical locations are not different, so the configuration of the phone is not modified.
B. The physical locations are different, so the roaming-sensitive parameters of the roaming device pool are applied.
C. The device mobility groups are the same, so the Device Mobility-related settings are applied in addition to the roaming-sensitive parameters.
D. The Device Mobility information is associated with one or more device pools other than the home device pool of the phone, so one of the associated device pools is chosen based on a round-robin load-sharing algorithm.
E. The Device Mobility information is associated with the home device pool of the phone, so the phone is considered to be in its home location. Device Mobility will reconfigure the roaming-sensitive settings of the phone.

Correct Answer: BC
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

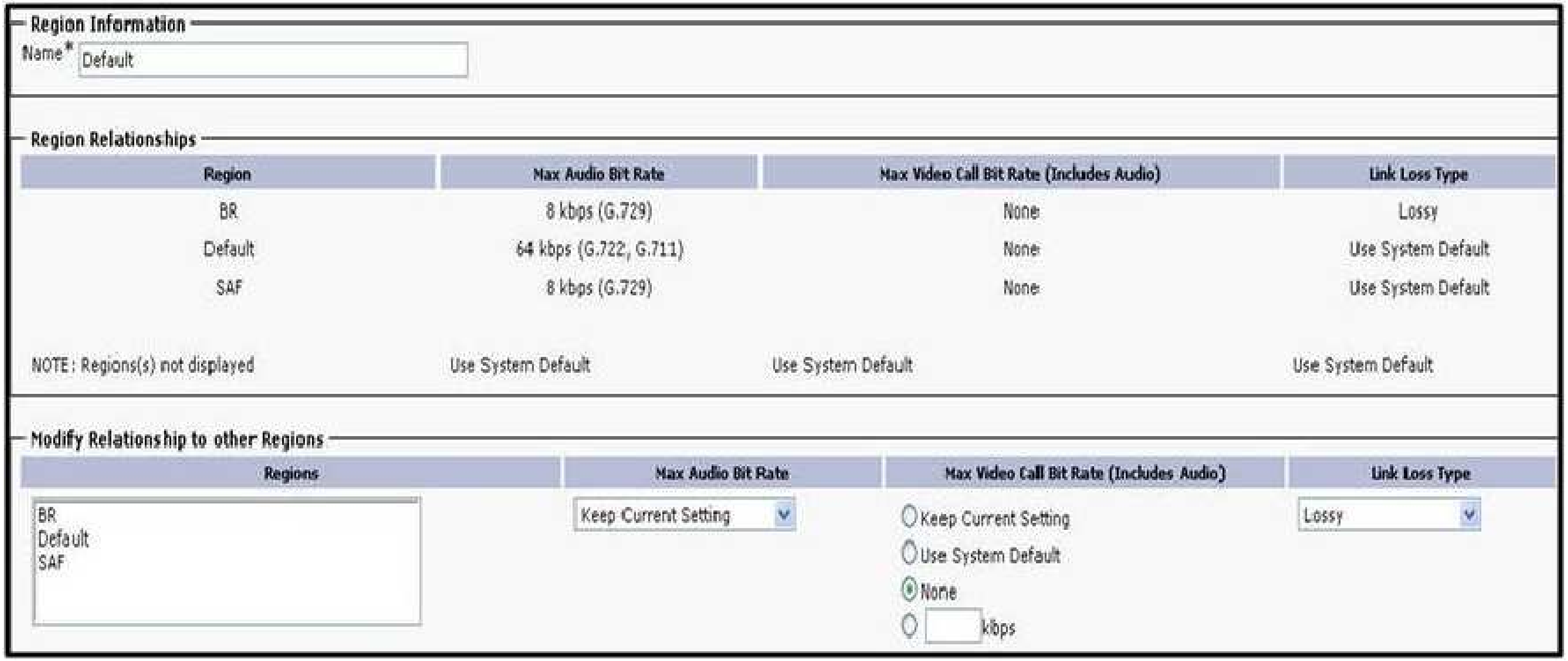
QUESTION 118
In a Centralized Call processing architecture, you have deployed Extension Mobility (EM) feature. After the deployment of EM, when one of the end-users tries to login to the IP phone, the Error 25 is displayed on the screen. What three things should you do to resolve this issue? (Choose three.)

A. upgrade the firmware of the IP Phone to the latest version
B. activate EM feature service under Cisco Unified Serviceability
C. associate EM Device profile with the end-user
D. subscribe the MAC address of the IP Phone to EM Service
E. update EM Phone Service URL to point to the publisher
F. subscribe device profile to EM phone service in case the enterprise subscription of EM Service is disabled

Correct Answer: BCD
Section: Implement Video Mobility Features
Explanation

Explanation/Reference:

QUESTION 119
Refer to the exhibit.



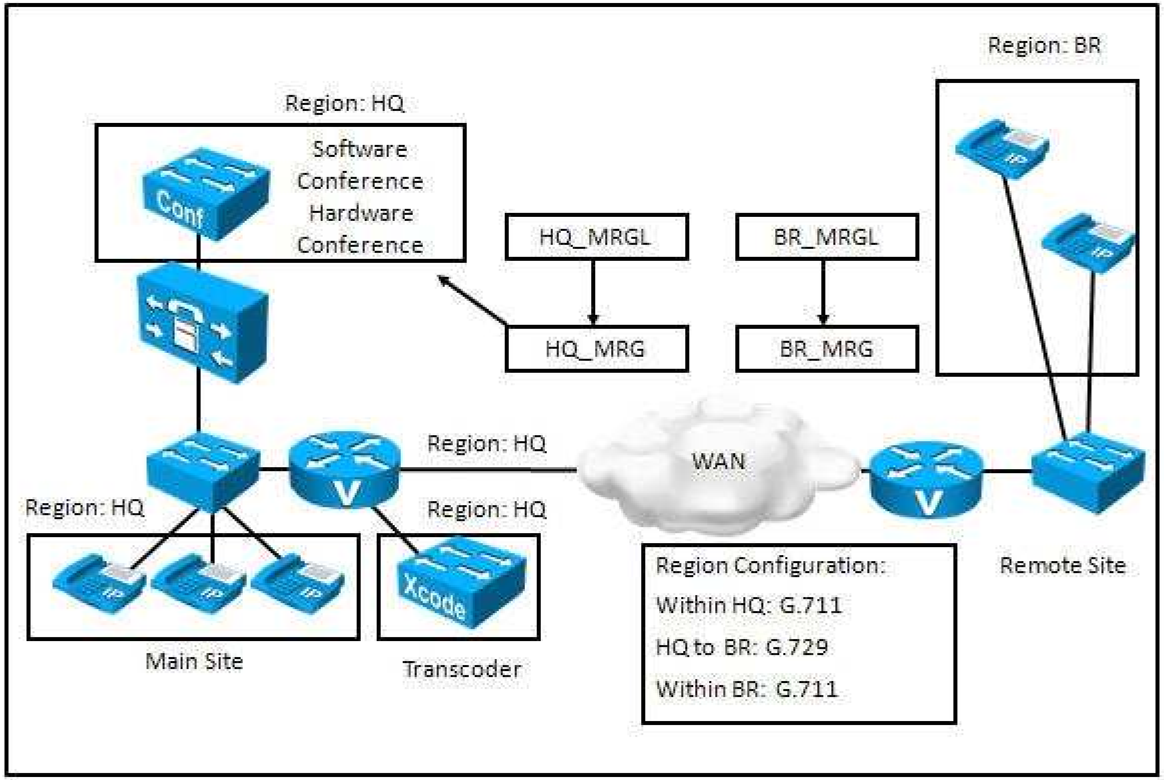
Which statement about the configuration between the Default and BR regions is true?

A. Calls between the two regions can use either 64 kbps or 8 kbps.
B. Calls between the two regions can use only the G.729 codec.
C. Only 64 kbps will be used between the two regions because the link is "lossy".
D. Both codecs can be used depending on the loss statistics of the link. When lossy conditions are high, the G.711 codec will be used.

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 120
Refer to the exhibit.



When a call between two HQ users is being conferenced with a remote user at BR, which configuration is needed?

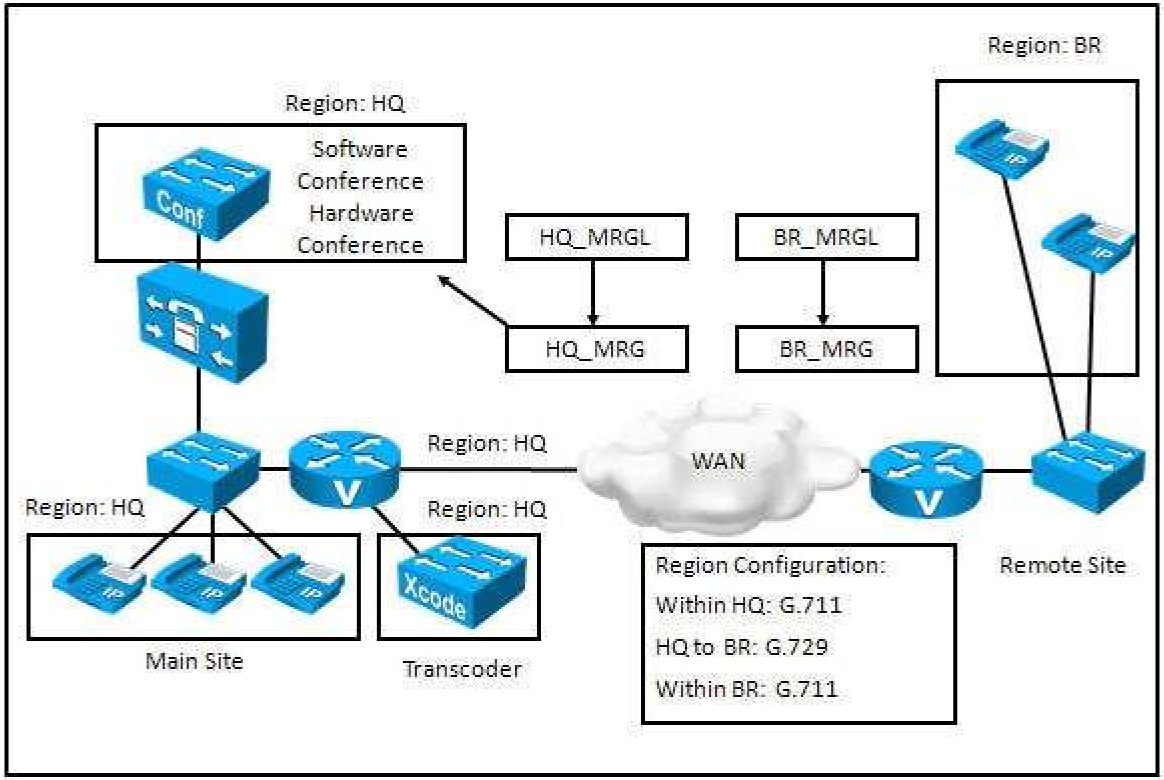
A. The BR_MRG must contain the transcoder device. The BR_MRGL must be assigned to the BR phones.
B. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the HQ phones.
C. A transcoder should be configured at the remote site and assigned to all remote phones through the BR_MRGL.
D. The HQ_MRG must contain the transcoder device. The HQ_MRGL must be assigned to the software conference bridge.
E. Enable the software conference bridge to support the G.711 and G.729 codecs in Cisco Unified Communications Manager Service Parameters.

Correct Answer: D

Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 121
Refer to the exhibit.



All HQ phones are configured to use HQ_MRGL and all BR phones are configured to use BR_MRGL. For the HQ phones always to use the hardware conference bridge as a first choice, which configuration should be implemented?

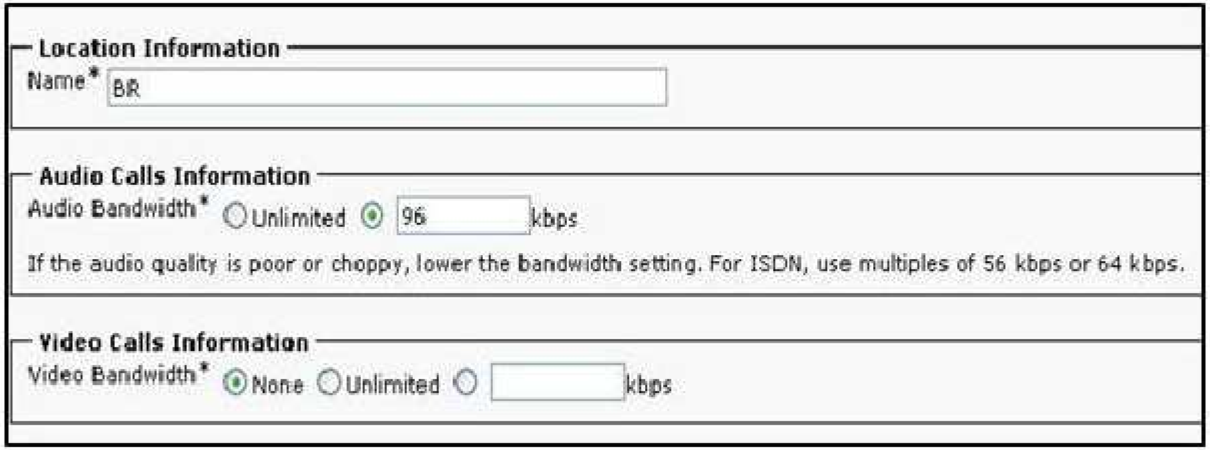
A. Ensure that both the hardware and software conference bridges are listed in the HQ_MRG. Ensure that the instance ID for the hardware conference bridge is 0.

B. Ensure that both the hardware and software conference bridges are listed in the HQ_MRG. The hardware conference bridge must be configured first.
C. Assign the hardware conference bridge to HQ_MRG. Configure a second HQ_MRG_2 and assign the software conference bridge to it. Add both the HQ_MRG and HQ_MRG_2 to the HQ_MRGL and list the HQ_MRG first.
D. Assign the hardware conference bridge to HQ_MRG. Configure a second HQ_MRG_2 and assign the software conference bridge to it. Configure an additional HQ_MRGL_2. Add the HQ_MRG to HQ_MRGL. Add HQ_MRG_2 to HQ_MRGL_2. The HQ_MRGL should be assigned to the HQ phones. The HQ_MRGL_2 should be assigned to the HQ device pool.

Correct Answer: C
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Expiation:
To ensure that the hardware bridge is utilized first with all its resources BEFORE the software bridge is used … you need to have two separate MRG's and list the hardware MRG 1st in the MRGL …

QUESTION 122
Refer to the exhibit.



Assuming the regions configuration to BR only permits G.729 codec, how many calls are allowed for the BR location?

A. Total of four calls; two incoming and two outgoing.

B. Total of two calls in either direction.

C. Total of four calls to the BR location. Outgoing calls are not impacted by the location configuration.

D. Total of four calls in either direction.

E. Two outgoing calls. Incoming calls are unlimited.

Correct Answer: D
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Incorrect answer: A, B, C, E
Explanation: In performing location bandwidth calculations for purposes of call admission control, Cisco Unified Communications Manager assumes that each G.729 call stream consumes 24 kb/s amount of bandwidth
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wpxref28640

QUESTION 123
Refer to the exhibit.

```
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 8.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ-1_MTP
!
dspfarm profile 1 mtp
 codec pass-through
 rsvp
 maximum sessions software 20
 associate application SCCP
!
interface Serial0/1
 description IP-WAN
 ip address 10.1.4.101 255.255.255.0
 duplex auto
 speed auto
 ip rsvp bandwidth 64
!
```

How many calls are permitted by the RSVP configuration?

A. one G.711 call
B. two G.729 calls
C. one G.729 call and one G.711 call
D. eight G.729 calls
E. four G.729 calls

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Incorrect answer: A, C, D, E
Explanation: In performing location bandwidth calculations for purposes of call admission control, Cisco Unified Communications Manager assumes that each call stream consumes the following amount of bandwidth:

- G.711 call uses 80 kb/s.
- G.722 call uses 80 kb/s.
- G.723 call uses 24 kb/s.
- G.728 call uses 26.66 kb/s.
- G.729 call uses 24 kb/s.

Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wpxref28640

QUESTION 124
Refer to the exhibit.

```
!
sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 8.0
sccp
!
sccp ccm group 1
 associate ccm 1 priority 1
 associate profile 1 register HQ-1_MTP
!
dspfarm profile 1 mtp
 codec pass-through
 rsvp
 maximum sessions software 20
 associate application SCCP
!
interface Serial0/1
 description IP-WAN
 ip address 10.1.4.101 255.255.255.0
 duplex auto
 speed auto
 ip rsvp bandwidth <value>
!
```

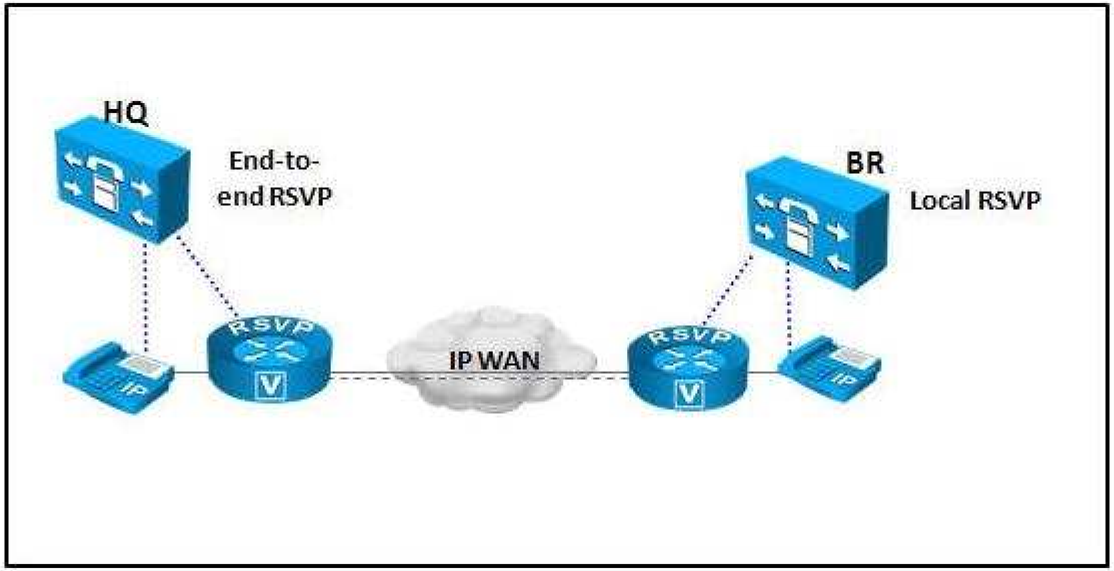
To permit three G.729 calls, what should the bandwidth value be for the ip rsvp bandwidth command?

A. 32
B. 48
C. 64
D. 88
E. 128

Correct Answer: D
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 125
Refer to the exhibit.



The HQ Cisco Unified Communications Manager has been configured for end-to-end RSVP. The BR Cisco Unified Communications Manager has been configured for local RSVP.

RSVP between the locations assigned to the IP phones and SIP trunks at each site are configured with mandatory RSVP. When a call is placed from the IP phone at HQ to the BR phone at the BR site, which statement is true?

A. The Cisco Unified Communications Manager at HQ will fall back to local RSVP and place the call. No RSVP end-to-end will occur.
B. RSVP end-to-end will occur.
C. The Cisco Unified Communications Manager at HQ will use end-to-end RSVP. The BR Cisco Unified Communications Manager will use local RSVP.
D. The call will fail.
E. The call will proceed as a normal call with no RSVP reservation.

Correct Answer: D
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Incorrect answer: A, B, C
Explanation: A possible cause is that the same router is being used as the calling and called RSVP agents, and that router is not running the latest IOS version,

which supports loopback on RSVP reservation. Make sure that the router is running the latest IOS version.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02rsvp.html#wp1155102

QUESTION 126
Which statement about H.323 Gatekeeper Call Admission Control is true?

A. Gatekeeper Call Admission Control applies to centralized Cisco Unified Communications deployments that use locations based Call Admission Control.
B. Gatekeeper Call Admission Control applies to distributed Cisco Unified Communications deployments.
C. Gatekeeper Call Admission Control applies only to distributed Cisco Unified Communications Express deployments.
D. Gatekeeper Call Admission Control setting is configured in Cisco Unified Communications Manager.

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Incorrect answer: A, C, D
Explanation: in distributed call processing deployments on a simple hub-and-spoke topology, you can implement call admission control with a Cisco IOS gatekeeper. In this design, the call processing agent (which could be a Unified CM cluster, Cisco Unified Communications Manager Express (Unified CME), or an H.323 gateway) registers with the Cisco IOS gatekeeper and queries it each time the agent wants to place an IP WAN call.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/srnd/7x/cac.html#wp1044743

QUESTION 127
Refer to the exhibit.

```
gatekeeper
zone local ClusterA lab.com 192.168.3.254
zone local ClusterB lab.com
zone prefix ClusterA 511*
zone prefix ClusterA 521*
zone prefix ClusterB 512*
zone prefix ClusterB 522*
bandwidth interzone default 64
bandwidth interzone zone ClusterB 48
gw-type-prefix 1#* default-technology
no shutdown
```

How many calls can be placed to Cluster B?

A. three G.729 calls
B. one G.711 call
C. one G.711 and three G.729 calls
D. There is no limit for incoming calls to Cluster B. Outgoing calls are limited to one G.711 and three G.729 calls.

Correct Answer: A
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 128
If your IP telephony administrator asks you to configure a local zone for your dial plan to control the volume of calls between two end points in a centralized multisite environment, which two types of Call Admission Control can be implemented? (Choose two.)

A. locations based
B. automated alternate routing
C. gatekeeper based
D. SRST
E. Cisco Unified Communications Manager based

Correct Answer: AB
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Incorrect answer: C, D, E
Explanation: Location-based call admission control (CAC) manages WAN link bandwidth in Cisco Unified Communications Manager. Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wp1067747

QUESTION 129
Refer to the exhibit. You have configured transcoder resources in both an IOS router and a Cisco Unified Communications Manager. When you review the configurations in both devices the IP addresses and transcoder names are correct, but the transcoder is failing to register with the Cisco Unified Communications Manager. Which command needs to be edited to allow the transcoder to register properly?

```
voice-card 0
 dspfarm
 dsp services dspfarm

sccp local FastEthernet0/0
sccp ccm 10.1.1.1 identifier 1 version 6.0
sccp

sccp ccm group 1
 associate ccm 2 priority 1
 associate profile 1 register HQ_XCODER

dsp farm profile 1 transcode
 codec g711ulaw
 codec g711alaw
 codec g729ar8
 codec g729abr8
 maximum sessions 2
 associate application SCCP
 no shutdown
```

A. The associate profile and dsp farm profile numbers need to match associate ccm 2 command.
B. The associate ccm 2 priority 1 command needs to be changed so the ccm value matches identifier 1 in the sccp ccm 10.1.1.1 command.
C. The sccp ccm group number needs to match the associate ccm 2 command.
D. The maximum sessions command must match the number of codecs configured under the dsp farm profile.
E. The sccp ccm group number must match the voice-card number.

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Incorrect answer: A, C, D, E
Explanation: The value of the IP address should match the IP address in the ip source-address command
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucme/admin/configuration/guide/cmevtrns.html

QUESTION 130
Which statement is correct about AAR?

A. The end users see, "Network Congestion Rerouting?" but AAR is otherwise transparent to the end user and works without user intervention.
B. AAR will display "not enough bandwidth" on the IP phone while it reroutes the call.
C. AAR allows calls to be rerouted because of insufficient Cisco Unified Border Element controlled bandwidth to an ITSP.
D. AAR allows calls to be rerouted due to insufficient gatekeeper controlled IP WAN bandwidth.

Correct Answer: A
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Incorrect answer: B, C, D
Explanation: Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth. With automated alternate routing, the caller does not need to hang up and redial the called party.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b03aar.html

QUESTION 131
The relationship between a Region and a Location is that the Region codec parameter is used between a Region and its configured Locations.

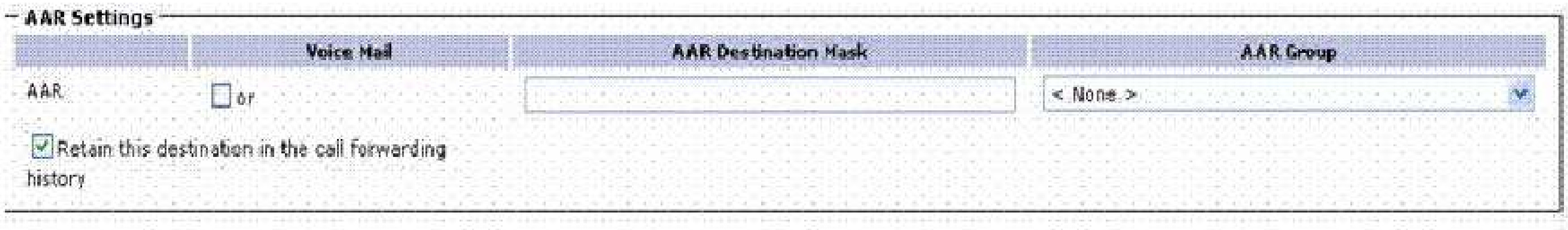
A. TRUE
B. FALSE

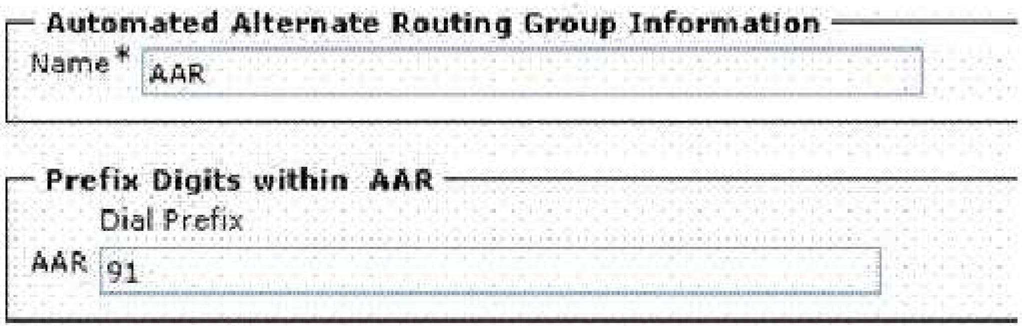
Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Explanation: Locations work in conjunction with regions to define the characteristics of a network link. Regions define the type of compression (G.711, G.722, G.723, G.729, GSM, or wideband) that is used on the link, and locations define the amount of available bandwidth for the link
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wp1033331

QUESTION 132

Refer to the exhibit. A user in RTP calls a phone in San Jose during congestion with Call Forward No Bandwidth (CFNB) configured to reach cell phone 4085550150. The user in RTP sees the message "Not Enough Bandwidth" on their phone and hears a fast busy tone. Which two conditions can correct this issue? (Choose two.)





A. The calling phone (RTP) needs to have AAR Group value of AAR under the AAR Settings.
B. The called phone (San Jose) needs to have AAR Group value of AAR under the AAR Settings.
C. The calling phone (RTP) needs to have the AAR destination mask of 914085550150 configured under the AAR Settings.
D. The calling phone (RTP) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.
E. The called phone (San Jose) needs to have the AAR destination mask of 914085550150 configured under the AAR Settings.
F. The called phone (San Jose) needs to have the AAR destination mask of 4085550150 configured under the AAR Settings.

Correct Answer: BF
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Incorrect answer: A, C, D, E

Explanation: Explanation: Automated alternate routing (AAR) provides a mechanism to reroute calls through the PSTN or other network by using an alternate number when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth. With automated alternate routing, the caller does not need to hang up and redial the called party.
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucmbe/admin/8_6_1/ccmcfg/b03aar.html

QUESTION 133
Which two statements describe RSVP-enabled locations-based CAC? (Choose two.)

A. RSVP can be enabled selectively between pairs of locations.
B. Using RSVP for CAC simply allows admitting or denying calls based on a logical configuration that is ignoring the physical topology.
C. RSVP is topology aware, but only works with full mesh networks.
D. An RSVP agent is a Media Termination Point that the call has to flow through.
E. RSVP and RTP are used between the two endpoints.

Correct Answer: AD
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Incorrect Answer: B, C
Explanation: The RSVP policy that is configured for a location pair overrides the default interlocation RSVP policy that configure in the Service Parameter Configuration window. RSVP supports audio, video, and data pass-through. Video data pass-through allows video and data packets to flow through RSVP agent and media termination point devices
Link: https://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02rsvp.html#wp1070214

QUESTION 134
The relationship between a Region and a Location is that the Region codec parameter is combined with Location bandwidth when communicating with other Regions.

A. FALSE
B. TRUE

Correct Answer: A
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:
Explanation: Locations work in conjunction with regions to define the characteristics of a network link. Regions define the type of compression (G.711, G.722, G.723, G.729, GSM, or wideband) that is used on the link, and locations define the amount of available bandwidth for the link
Link: http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/8_6_1/ccmsys/a02cac.html#wp1033331

QUESTION 135
You are deploying a remote office setup that connects with Cisco Unity Communications Manager at a hub location. You have an available dedicated bandwidth of 20% from the 2-Mb/s WAN circuit for VoIP that supports a maximum of 17 calls. Which codec do you configure in Cisco Unity Communications Manager to achieve this?

A. G.722
B. G.711
C. G.729
D. iSAC
E. GSM-FR
F. iLBC

Correct Answer: C
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 136
Which task must you perform before deleting a transcoder?

A. Delete the dependency records.
B. Unassign it from a media resource group.
C. Use the Reset option.
D. Remove the device pool.
E. Remove the subunit.
F. Delete the common device configuration.

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 137
Which two are gatekeeper-controlled trunk options that support gatekeeper call administration control? (Choose two.)

A. H.323
B. H.245
C. H.225
D. intercluster
E. intracluster

Correct Answer: CD
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 138
Which action configures PSTN backup for calls that are rejected by the gatekeeper CAC?

A. Configure AAR in Cisco Unified Communications Manager.
B. Configure CFUR in Cisco Unified Communications Manager.
C. Configure a route pattern, a route list, and route groups to a trunk and a gateway in Cisco Unified Communications Manager.
D. Configure a route pattern to a gateway in Cisco Unified Communications Manager.

Correct Answer: C
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 139
Cisco Unified Communications Manager is configured with CAC for a maximum of 10 voice calls.

Which action routes the 11th call through the PSTN?

A. Configure an SIP trunk to the ISR.
B. Configure Cisco Unified Communications Manager AAR.
C. Configure Cisco Unified Communications Manager RSVP-enabled locations.
D. Configure Cisco Unified Communications Manager locations.

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 140
How do RSVP-enabled locations differ from Cisco Unified Communications Manager locations?

A. RSVP is configured in the ISR independent of Cisco Unified Communications Manager.
B. RSVP enables AAR within Cisco Unified Communications Manager.
C. RSVP is topology aware.
D. RSVP is configured in Cisco Unified Communications Manager independent of the ISR.

Correct Answer: C
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 141
A voice-mail product that supports only the G.711 codec is installed in headquarters.

Which action allows branch Cisco IP phones to function with voice mail while using only the G.729 codec over the WAN link to headquarters?

A. Configure Cisco Unified Communications Manager regions.
B. Configure transcoding within Cisco Unified Communications Manager.
C. Configure transcoding resources in Cisco IOS and assign to the MRGL of Cisco IP phones.
D. Configure transcoder resources in the branch Cisco IP phones.

Correct Answer: C
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 142
Which system configuration is used to set a restriction on audio bandwidth?

A. region
B. location
C. physical location
D. licensing

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 143
What happens if location-based CAC is used and there is no bandwidth available when a remote caller is placed on hold?

A. Cisco Unified Communications Manager sends TOH rather than MOH.
B. Cisco Unified Communications Manager terminates the call.
C. Cisco Unified Communications Manager plays default MOH.
D. Cisco Unified Communications Manager attempts to reconnect the call immediately.

Correct Answer: A
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 144
On which Cisco Unified Communications Manager configuration parameter does the CODEC that a Cisco IP Phone uses for a call depend?

A. enterprise parameters
B. media resources
C. physical location
D. region

E. location

Correct Answer: D
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 145
In a Cisco Unified Communications Manager centralized call processing model, what is the best CAC method recommended for this type of deployment?

A. QoS-based
B. location-based
C. RSVP-based
D. region-based
E. gateway-based
F. gatekeeper-based

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 146
Which action configures phones in site A to use G.711 to site B but to use G.729 to site C?

A. Configure Cisco Unified Communications Manager regions.
B. Configure Cisco Unified Communications Manager locations.
C. Configure transcoder resources in Cisco Unified Communications Manager.
D. Configure a gatekeeper.

Correct Answer: A
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 147
Which statement is true when considering a Cisco VoIP environment for regional configuration?

A. G.711 requires 128K of bandwidth per call.
B. G.729 requires 24K of bandwidth per call.
C. The default codec does not matter if you have defined a hardware MTP in your Cisco Unified Communications Manager environment.
D. To deploy a Cisco H.323 gatekeeper, you must configure MTP resources on the gatekeeper and only use G.711 between regions.

Correct Answer: C
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 148
Which system configuration is used to set audio codecs?

A. region
B. location
C. physical location
D. licensing

Correct Answer: A
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 149
When you configure regions in a Cisco Unified Communications Manager multisite cluster, which two are ways to prevent G.722 from being advertised in the cluster? (Choose two.)

A. modify the service parameter
B. modify the enterprise parameter
C. modify the device pool

D. modify the line settings

Correct Answer: AB
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 150
Which action configures the registration of transcoder resources?

A. Cisco IOS software registers transcoder resources with SIP.
B. Cisco IOS software registers transcoder resources with SCCP.
C. Cisco IOS software registers transcoder resources with H.323.
D. Cisco IOS software does not register transcoder resources.

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 151
Which option describes the reason that transcoding resources are added in Cisco Unified Communications Manager?

A. to enable transcoding resources in a Cisco Unified Communications Manager server
B. to enable Cisco Unified Communications Manager to select the optimal single codec for end-to-end calls
C. to enable transcoding resources in Cisco IP Phones
D. to provide transcoding resources in Cisco IOS gateways to Cisco Unified Communications Manager

Correct Answer: D
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 152
Which action configures transcoding resources in Cisco Unified Communications Manager to function with branch office Cisco IP Phones?

A. Configure the branch office IP phones with CSS and partitions.
B. Configure the branch office IP phones with MRGs and MRGLs.
C. Configure the branch office IP phones with regions.
D. Configure the branch office IP phones with locations.

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 153
What is the purpose of configuring a hardware-based MTP when deploying Cisco Unified Communications Manager?

A. to allow for supplementary services such as hold, transfer, and conferencing
B. when you need support for up to 24 MTP sessions on the same server and 48 on a separate server
C. when you need the ability to grow support by using DSPs
D. when you want to only use Cisco Unified Communications Manager resources

Correct Answer: C
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 154

Which two options are effective mechanisms to restrict the maximum number of voice calls on a WAN link? (Choose two.)

A. Configure a gatekeeper with an SIP trunk.
B. Configure a gatekeeper and a gatekeeper-controlled trunk in Cisco Unified Communications Manager with bandwidth control.
C. Configure Cisco Unified Communications Manager regions.
D. Configure Cisco Unified Communications Manager locations.

Correct Answer: BD
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 155
Which action configures CAC utilizing only Cisco Unified Communications Manager software?

A. Configure Cisco Unified Communications Manager regions.
B. Configure Cisco Unified Communications Manager locations.
C. Configure Cisco Unified Communications Manager RSVP-enabled locations.
D. Configure Cisco Unified Communications Manager MTPs.

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 156
Which bandwidth amounts are correct for configuring locations?

A. 8 kb/s for G.729, 64 kb/s for G.711, and 64 kb/s for G.722
B. 8 kb/s for G.729, 64 kb/s for G.711, and 16 kb/s for G.722
C. 64 kb/s for G.729, 64 kb/s for G.711, and 64 kb/s for G.722
D. 8 kb/s for G.729, 8 kb/s for G.711, and 8 kb/s for G.722

Correct Answer: A

Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 157
Which action configures AAR to route the calls that have been rejected by the gatekeeper CAC through the PSTN?

A. Configure Cisco IP Phones for AAR.
B. Configure AAR to work with SRST.
C. Configure AAR to work with CTI route points.
D. This configuration is not possible using AAR.

Correct Answer: D
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 158
What is a prerequisite of AAR deployment?

A. You must have a single distributed call processing deployment.
B. Calls must be manually rerouted through the PSTN or other networks when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth.
C. Calls must be automatically rerouted through the PSTN or other networks when Cisco Unified Communications Manager blocks a call due to insufficient location bandwidth.
D. Clustering must be implemented over the WAN.
E. You must have a centralized call processing deployment.

Correct Answer: E
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 159
How is the region assigned to a device such as an IP phone?

A. Regions are assigned directly in the device configuration page.
B. Regions can be assigned only through a device pool.
C. Regions can be assigned either directly on the device configuration page or through the device pool. If both configurations exist, the device pool region configuration takes precedence.
D. Regions can be assigned either directly on the device configuration page or through the device pool. If both configurations exist, the device region configuration takes precedence.

Correct Answer: B
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 160
How can the location setting be modified to resolve poor call quality?

A. No adjustment to location setting is needed
B. Mark the bandwidth between the locations as unlimited
C. Decrease the audio bandwidth setting
D. Remove the audio bandwidth setting

Correct Answer: C
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 161
Cisco Unified border element is configured to support RSVP-based CAC. When is the RSVP path and reservation message sent and received?

A. Immediately after the call setup message is received and the reservation message is received after H.245 capabilities negotiation is completed.
B. The path and reservation messages are sent and received after the H.245 capabilities negotiation is completed.
C. The path and reservation messages are sent and received immediately after the call setup message is received.
D. The path is setup once the global command call rsvp-sync is configured.

Correct Answer: C
Section: Implement Bandwidth Management and Call Admission Control on CUCM
Explanation

Explanation/Reference:

QUESTION 162
Company A has deployed a VCS Control and is attempting to register a third-party endpoint. The engineer has confirmed that no traffic is being blocked for the endpoint and it is receiving a valid IP address. Which option could be the cause of this registration failure?

A. Third-party endpoints are not compatible with VCS Control, only with VCS Expressway.
B. Cisco Unified Communications Manager is required in addition to the VCS Control.
C. An incorrect SIP domain is configured on the VCS Control for the endpoint.
D. The VCS Control must be deployed together with VCS Expressway before endpoints can register to either one.

Correct Answer: C
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 163
Company X has deployed a VCS Control with a local zone and a traversal client zone. To facilitate external calls, VCS Expressway is deployed and traversal server zone is set up there. Video endpoints inside Company X have registered but are unable to receive calls from outside endpoints. Which option could be the cause of this issue?

A. The access control list on the VCS Control must be updated with the IP for the external users.
B. When a traversal zone is set up on VCS Control only outbound calls are possible.
C. The local zone on the VCS Control does not have a search rule configured.
D. The traversal zone on the VCS Control does not have a search rule configured.

Correct Answer: C
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 164
When a call is made from a video endpoint to a Cisco TelePresence EX90 that is registered to a Cisco VCS Control, which portion of the destination URI is the first match that is attempted?

A. the full URI, including the domain portion
B. the destination alias, without the domain portion
C. the E.164 number that is assigned to the Cisco TelePresence EX90
D. the directory number that is assigned to the Cisco TelePresence EX90

Correct Answer: B
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 165
Which two options are requirements for deploying an H.323 gateway with Cisco Unified Communications Manager? (Choose two.)

A. Cisco Unified Communications Manager and the H.323 gateway must be configured use the same TCP port for H.323 calls.
B. The H.245TCSTimeout timer must be set to at least 25.
C. Cisco voicemail ports must be active.
D. The Media Exchange Interface Capability Timer must be set to less than 20.
E. The Media Exchange Timer must be set to less than 20.

Correct Answer: AB
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 166
Which two configurations can you perform to allow Cisco Unified Communications Manager SIP trunks to send an offer in the INVITE? (Choose two.)

A. Enable the Media Termination Point Required option on the SIP trunk.
B. Enable the Early Offer Support for Voice and Video Calls option on the SIP profile.

C. Select the Display IE Delivery check box in the gateway configuration.
D. Select the Enable Inbound FastStart check box on the Cisco Unified Communications Manager servers.
E. Select the SRTP Allowed check box on the SIP trunk.
F. Execute the isdn switch-type primary-ni command globally.

Correct Answer: AB
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 167
When configuring a video ISDN gateway, which two actions are requirements for the Cisco Preferred Architecture for Enterprise Collaboration? (Choose two.)

A. Use SIP instead of H.323.
B. Perform dial string manipulation on Cisco Unified Communications Manager.
C. Use an * (asterisk) at the end of each ISDN number, as a suffix.
D. Use an ! (exclamation point) at the end of each ISDN number, as a suffix.
E. Use H.323 instead of SIP.

Correct Answer: AB
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 168
Which two actions ensure that the call load from Cisco TelePresence Video Communication Server to a Cisco Unified Communications Manager cluster is shared across Unified CM nodes? (Choose two.)

A. Create a neighbor zone in VCS with the Unified CM nodes listed as location peer addresses.
B. Create a single traversal client zone in VCS with the Unified CM nodes listed as location peer addresses.
C. Create one neighbor zone in VCS for each Unified CM node.
D. Create a VCS DNS zone and configure one DNS SRV record per Unified CM node.
E. In VCS set Unified Communications mode to Mobile and remote access and configure each Unified CM node.

Correct Answer: AD
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 169
Which two options must be selected in the SIP Trunk Security Profile configuration between Cisco Unified Communications Manager and Expressway? (Choose two)

A. Enable application-level authorization
B. Accept presence subscription
C. Accept out-of-dialog refer
D. Accept unsolicited notification
E. Accept replaces header
F. Transmit security status
G. Allow charging header

Correct Answer: DE
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 170
Which device must be accessible from the public Internet in a Collaboration Edge environment?

A. Expressway-C
B. Cisco Unified Communications Manager
C. Cisco IM and P
D. Expressway-E
E. VCS Control

Correct Answer: D
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 171
You are deploying a Cisco Unified Communications Manager solution with MGCP gateways at multiple locations. Which firewall and ACL configuration must you perform to allow the MCGP gateways to function correctly?

A. Allow access to TCP port 2428.
B. Block TCP port 1720.
C. Open access to all TCP and UDP ports.
D. Allow access to TCP port 1720.
E. Block access to TCP ports 2427 and 2428.

Correct Answer: A
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 172
The VCS Expressway can be configured with security controls to safeguard external calls and endpoints. One such option is the control of trusted endpoints via a whitelist. Where is this option enabled?

A. on the voice-enabled firewall at the edge of the network
B. on the VCS under Configuration > registration > configuration
C. on the TMS server under Registrations > whitelist
D. on the VCS under System > configuration > Registrations

Correct Answer: B
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 173
On which two call legs is the media encryption enforced in a Collaboration Edge design? (Choose two.)

A. Expressway-C to Cisco Unified Communications Manager
B. Expressway-C to Expressway-E
C. Expressway-E to outside-located endpoint
D. Expressway-E to Cisco Unified Communications Manager
E. Expressway-C to internal endpoint

Correct Answer: BC
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 174
Which two statements regarding you configuring a traversal server and traversal client relationship are true? (Choose two.)

A. VCS supports only the H.460.18/19 protocol for H.323 traversal calls.
B. VCS supports either the Assent or the H.460.18/19 protocol for H.323 traversal calls.
C. VCS supports either the Assent or the H.460.18/19 protocol for SIP traversal calls.
D. If the Assent protocol is configured, a TCP/TLS connection is established from the traversal client to the traversal server for SIP signaling.
E. A VCS Expressway located in the public network or DMZ acts as the firewall traversal client.

Correct Answer: BD
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 175
What is the standard Layer 3 DSCP media packet value that should be set for Cisco TelePresence endpoints?

A. CS3 (24)
B. EF (46)
C. AF41 (34)
D. CS4 (32)

Correct Answer: D
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 176
When you configure QoS on VCS, which settings do you apply if traffic through the VCS should be tagged with DSCP AF41?

A. Set QoS mode to DiffServ and tag value 32.
B. Set QoS mode to IntServ and tag value to 34.
C. Set QoS mode to DiffServ and tag value 34.
D. Set QoS mode to IntServ and tag value to 32.
E. Set QoS mode to ToS and tag value to 32.

Correct Answer: C
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 177
What is the default DSCP/PHB for TelePresence video conferencing packets in Cisco Unified Communications Manager?

A. EF/46
B. CS6/48
C. AF41/34
D. CS3/24
E. CS4/32

Correct Answer: E
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 178
Which three commands are necessary to override the default CoS to DSCP mapping on interface Fastethernet0/1? (Choose three.)

A. mls qos map cos-dscp 0 10 18 26 34 46 48 56
B. mls qos map dscp-cos 8 10 to 2
C. mls qos
D. interface Fastethernet0/1
   mls qos trust cos
E. interface Fastethernet0/1
   mls qos cos 1
F. interface Fastethernet0/2
   mls qos cos 1

Correct Answer: ACD
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 179
Which option indicates the best QoS parameters for interactive video?

A. 1% Max Loss, 150 ms One-way Latency, 30 ms Jitter, 20% Overprovisioning
B. 5% Max Loss, 5 s One-way Latency, 30 ms Jitter, 20% Overprovisioning
C. 0% Max Loss, 100 ms One-way Latency, 30 ms Jitter, 20% Overprovisioning
D. 1% Max Loss, 160 ms One-way Latency, 60 ms Jitter, 10% Overprovisioning

Correct Answer: A
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 180
Which two options are valid service parameter settings that are used to set up proper video QoS behavior across the Cisco Unified Communications Manager infrastructure? (Choose two.)

A. DSCP for Video Calls when RSVP Fails
B. Default Intraregion Min Video Call Bit Rate (Includes Audio)
C. Default Interregion Max Video Call Bit Rate (Includes Audio)
D. DSCP for Video Signaling
E. DSCP for Video Signaling when RSVP Fails

Correct Answer: AC
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 181
Refer to the exhibit.

```
Router#sh policy-map interface serial0/3/0
 Serial0/3/0

  Service-policy output: VOICE-VIDEO

    queue stats for all priority classes:

      queue limit 64 packets
      (queue depth/total drops/no-buffer drops) 0/0/0
      (pkts output/bytes output) 0/0

    Class-map: VOICE (match-all)
      0 packets, 0 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match:  dscp ef (46)
      Priority: 10% (153 kbps), burst bytes 3800, b/w exceed drops: 0


    Class-map: VIDEO (match-all)
      0 packets, 0 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match:  dscp af41 (34)
      Queueing
      queue limit 64 packets
      (queue depth/total drops/no-buffer drops) 0/0/0
      (pkts output/bytes output) 0/0
      bandwidth 25% (384 kbps)

    Class-map: TELEPRESENCE (match-all)
      0 packets, 0 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match:  dscp af32 (28)
      Queueing
      queue limit 64 packets
      (queue depth/total drops/no-buffer drops) 0/0/0
      (pkts output/bytes output) 0/0
      bandwidth 25% (384 kbps)

    Class-map: class-default (match-any)
      10 packets, 560 bytes
      5 minute offered rate 0 bps, drop rate 0 bps
      Match: any
      Queueing
      queue limit 64 packets
      (queue depth/total drops/no-buffer drops/flowdrops) 0/0/0/0
      (pkts output/bytes output) 10/560
      Fair-queue: per-flow queue limit 16
```

What is the correct value to use for the "DSCP for TelePresence Calls" Cisco CallManager service parameter?

A. 28 (011100)
B. 34 (100010)
C. 41 (101001)
D. 46 (101110)

Correct Answer: A
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 182
In the case of a VCS cluster, which configuration element is recommended for endpoint H.323 registration?

A. DNS SRV records pointing to the VCS cluster name
B. static IP addresses
C. hostname of the VCS cluster configuration master
D. hostname of the VCS cluster member peer

Correct Answer: A
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 183
In a distributed call processing network with locations-based CAC, calls are routed to and from intercluster trunks. Which trunk type is implemented in this network?

A. intercluster trunk with gatekeeper control
B. intercluster trunk without gatekeeper control
C. SIP trunk
D. h225 trunk

Correct Answer: B

Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 184
Which two options should be selected in the SIP trunk security profile that affect the SIP trunk pointing to the VCS? (Choose two.)

A. Accept Unsolicited Notification
B. Enable Application Level Authorization
C. Accept Out-of-Dialog REFER
D. Accept Replaces Header
E. Accept Presence Subscription

Correct Answer: AD
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 185
Company X has a Cisco Unified Communications Manager cluster and a VCS Control server with video endpoints registered on both systems. Users find that video endpoints registered on Call manager can call each other and likewise for the endpoints registered on the VCS server. The administrator for Company X realizes he needs a SIP trunk between the two systems for any video endpoint to call any other video endpoint. Which two steps must the administrator take to add the SIP trunk? (Choose two.)

A. Set up a SIP trunk on Cisco UCM with the option Device-Trunk with destination address of the VCS server.
B. Set up a subzone on Cisco UCM with the peer address to the VCS cluster.
C. Set up a neighbor zone on the VCS server with the location of Cisco UCM using the menu option VCS Configuration > Zones > zone.
D. Set up a SIP trunk on the VCS server with the destination address of the Cisco UCM and Transport set to TCP.
E. Set up a traversal subzone on the VCS server to allow endpoints that are registered on Cisco UCM to communicate.

Correct Answer: AC
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 186
Which symbol is required for globalized call routing?

A. +
B. *
C. %
D. /
E. ;

Correct Answer: A
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 187
Which two statements about SAF service identifier numbers are true? (Choose two.)

A. They are generated in the format service:sub-service:instance.instance.instance.instance.
B. They are 16-bit decimal identifiers.
C. They are generated in the format data-source:sub-service:instance.matrix.fifty.saf.
D. They are 32-bit decimal identifiers.
E. They are generated in the format data.saf.cucm-publisher.asf@domain.local.
F. They are generated in the format telco.cisco.saf-forwader.db.replicate.data.local.

Correct Answer: AB
Section: Mixed Questions
Explanation

Explanation/Reference:

QUESTION 188
Which three options describe the main functions of SAF Clients? (Choose three.)

A. registering the router as a client with the SAF network

B. providing publishing services to the SAF network
C. subscribing to SAF network services
D. registering Cisco Unified Communications Manager subscribers with the publisher
E. starting Cisco Unified Communications Manager services throughout the cluster
F. integrating with Cisco IM and Presence for additional services

Correct Answer: ABC
Section: Mixed Questions
Explanation

Explanation/Reference: